Simulation Study of a Mobile Satellite System

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Certificate

It is certified that the work contained in the thesis entitled **Simulation Study of a Mobile Satellite System** by Mukul Katiyar has been carried out under my supervision and that this work has not been submitted elsewhere for a degree

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Abstract

Mobile communication networks encompass the terrestrial cellular networks and the satellite mobile networks. The recent advances in satellite technology have stimulated the development of *Universal Mobile Telecommunication systems*. These mobile satellite networks can provide global or regional coverage of urban as well as sparsely populated (oceanic) areas. This is in contrast to the cellular mobile systems, coverage for which is limited to densely populated urban areas.

In this thesis an attempt has been made to analyze the protocols and evaluate the performance characteristics for such a system, proposed by the Indian Space Research Organization (I.S.R.O.) and called as the **INSAT Mobile Satellite System**. Discrete time event type of simulation was adopted as the technique for analyzing the system.

Simulators were developed for the circuit switched **Class-A** type of system, providing voice, data and Fax services using *demand assigned* SCPC channels and the store and forward **Class B** type of system. Analysis is performed based on data generated from simulations carried out under different traffic conditions. Finally some modifications to the proposed schemes have been suggested based on the analysis.

Contents

List of Figures

List of Abbreviations

ACK	Acknowledgement
BB	Bulletin Board Packet
BCD	Binary Coded Decimal
BER	Bit Error Rate
CRC	Cyclic Redundancy Code
CW	Control Word
DTE	Data Terminal Equipment
FEC	Forward Error Correction
GEO	Geostationary Earth Orbit
I.S.R.O.	Indian Space Research Organisation
ISU	Initial Signal Unit
LCC	Lost Calls Clear
LEO	Low Earth Orbit
MEO	Medium Earth Orbit
MFR	Message Forward Request
MRA	Message Received Acknowledgement
MSS	Mobile Satellite System
MTR	Message Transfer Request
PSTN	Public Switched Telephone Network
SCPC	Single Channel Per Carrier
SSU	Subsequent Signal Unit
SU	Signal Unit
TDM	Time Division Multiplexed
UW	Unique Word

Chapter 1

INTRODUCTION

1.1 Introduction to Mobile Communication

The unique feature of communication satellites is their ability to simultaneously link all users on the earth's surface, thereby providing distance insensitive point to multipoint communications. This capability applies to fixed terminals on earth and to mobile terminals on land, in the air, and at sea. The satellite's, capacity can also be dynamically allocated to users who need it. These features, make satellite communications systems unique in design.

Although cellular terrestrial mobile communications systems exist, they only serve the urban areas where base stations are located relatively close to the mobile users. The cellular systems are not economically possible for vehicular communications in rural or remote areas where population density is low. Satellite technology has reached a point where vehicular communications between mobile users and base stations can be achieved with low cost. Such mobile satellite systems can complement existing cellular terrestrial mobile systems by extending the communication coverage from urban to rural areas. These systems are not restricted to land coverage but can include maritime services as well. Some applications for telephone and data communications with Mobile Satellite Systems include transportation (coastal marine, rail, school buses, public utilities), vehicle location and tracking, air telephone, land telephone, interactive data, public safety and medicine.

1.2 Mobile Satellite Networks Trends

The 1980s saw the introduction of first and second generation digital geostationary orbit (GEO) mobile satellite systems (MSS). The examples of this are the Inmarsat voice, data and facsimile services. This has continued into the 1990-95 time frame with the deployment of more advanced second generation GEO systems such as Inmarsat M which provides voice, data and facsimile services using digital modulation techniques to briefcase sized portable and mobile terminals. For the 1998+ time frame radically new mobile satellite systems have been proposed with even smaller spot beams to provide telephony to hand held terminals. Geostationary earth orbit (GEO 36,000 Km), medium earth orbit (MEO 10,500 Km) and low earth orbit (LEO, 700-1,600 Km) constellations of satellite have been proposed, including the GEO MEO Inmarsat P21, the MEO Odyssey, Globalstar and Aries and the LEO Iridium and Ellipso systems [2]. These systems will provide voice, data, facsimile and navigational services to small hand held terminal using digital modulation techniques. They are expected to provide global (or almost global) and regional coverage of poorly served or sparsely populated (e.g. oceanic) areas. They will support dual mode hand held terminals which will be able to connect to both a satellite system and the terrestrial cellular network. Whenever the hand held terminals are within the range of a terrestrial mobile network, this network will be used for transmission of information.

In the future, mobile satellite systems will provide effectively continuous global and regional coverage of the earth's surface using constellations of orbiting satellites [2]. Some of the mobile satellite systems will also be able to relay information through the satellite network by crosslinking satellites (intersatellite links) for communication between one part of the globe and another. In some cases (e.g. LEO systems) any satellite in the constellation will be able to communicate directly with any other system component on the earth's surface (i.e. without satellite earth stations) [2].

1 3 Motivation

In this thesis we have tried to study the protocols and signaling schemes for the services that may be provided by the Mobile Satellite System proposed by the Indian Space Research Organization (I S R O) This system will be operated through the INSAT 2C and INSAT 2D satellites

In order to study a system we often have to make a set of assumptions about how it works These assumptions which usually take the form of mathematical or logical relationships, constitute a model that is used to try to gain some understanding of how the corresponding system behaves

If the model is simple enough it may be possible to use mathematical methods to obtain exact information on questions of interest, this is called an analytic solution However, most real world systems are too complex to allow realistic models to be evaluated analytically, and these models must be studied by means of simulation In a simulation model we use a computer to evaluate a model numerically and data are gathered in order to estimate the desired true characteristics of the model Here we have made use of simulation to study the performance of the ISRO's Mobile Satellite System

1 4 System Overview

The INSAT Mobile Satellite System (INSAT MSS) consists of the three major components as shown in Fig 1 2 The mobile communication payloads on the INSAT 2C and 2D satellites form the first component of the system The mobile terminals constitute the second component The third component is the HUB It provides the gateway to terrestrial networks for communication between mobile users and fixed PSTN subscribers and network management functions These three components provide the communication system along with the necessary signaling and access control functions in addition to voice, data and other services

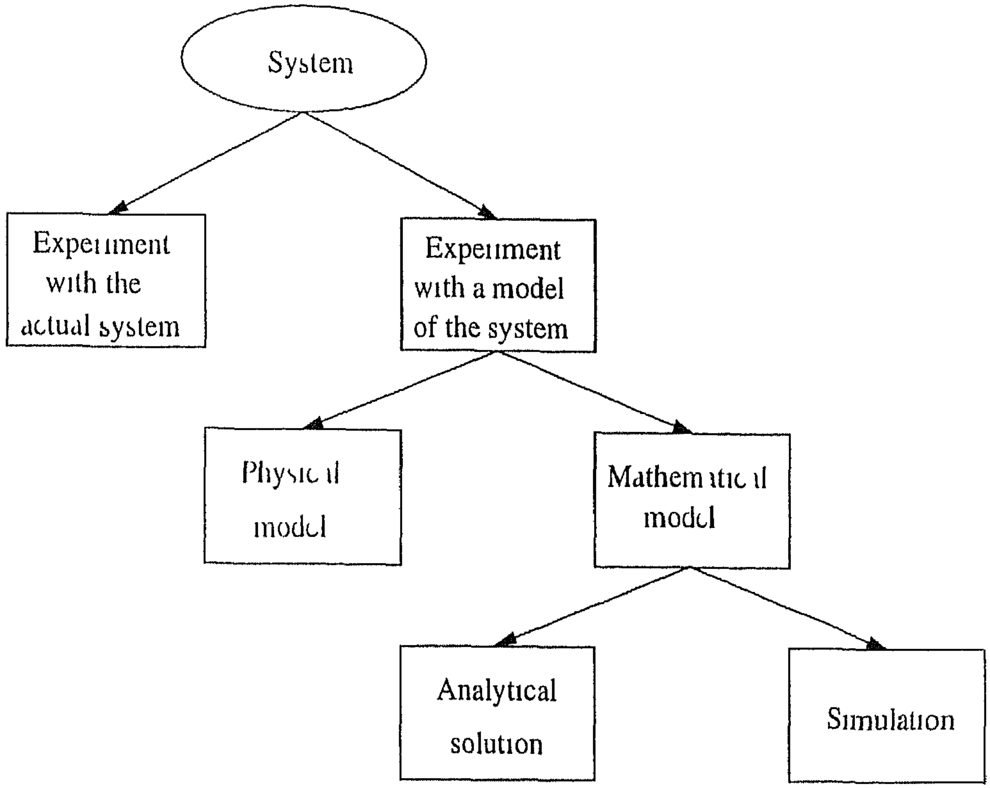


Figure 1.1 Various Methods of Analysis

1.4.1 INSAT-MSS Services

The services to be provided by INSAT MSS are classified as class-A, class B and class C as given below

- Class A Voice, data and fax using SCPC communication channels
- Class B Messaging using shared channels (Store and forward)
- Class C Reporting services

MSS will provide voice, data and fax services using demand assigned SCPC channels. These services are referred to as *Class A* services. Voice service is provided by using low bit rate voice coders which will provide communication quality speech. Data service provides end to end data circuits between a mobile DTE and a PSTN subscriber DTE at a data rate of 2400 bps. Facsimile service is provided at 2400 bps data rate between a PSTN subscriber fax and auto-answer Group 3 fax at mobile

terminal

The *Class B* type of service is a low bit rate store and forward messaging service using shared channels The HUB acts as the store and forward unit for this service The DTE in a mobile terminal will be equipped appropriately for message handling This service is distinct from Class A data service since no dedicated circuit is provided for messaging

The *Class C* service called reporting service will be available in simplex mode Here also the HUB acts as the store and forward unit for forwarding the reporting messages to either telex or PSTN circuits

All these services will be available for land mobile and maritime users Land mobile services will be restricted to the rural areas and open highways where the link availability is better than 90% Portable terminals can be operated in clear line of sight conditions by pointing the antenna towards the satellite

The following are the services to be provided by INSAT MSS

- Direct dial telephone interconnection between mobile terminals and PSTN subscribers

- Direct dial telephone interconnection between one mobile terminal and another mobile terminal

- Full duplex data interconnection between a mobile and a PSTN data circuit at 2400 bps data rate Mobile terminals will have an RS 232C/ V 24 interface and will emulate Hayes modem dialing responses Data transmission will be protected against short duration fades

- Direct dial and auto answer Group 3 fax interconnection between PSTN and mobile terminals Transmission rate will be 2400 bps Protection against short duration fades will be provided

- Voice data and fax group call services for mobile terminals in which a PSTN subscriber with appropriate authorization can broadcast to a group of mobile

users This facility will allow a single transmission to be received by a group of users

- Store and Forward Message transfer to telex network HUB acts as the store and forward node System assures complete and correct message transfers There is an end to end acknowledgement of successful message transfers
- Store and forward Message transfer to PSTN data lines
- Store and forward Message transfer to mobile terminals
- HUB will provide a Mail box service Messages which could not be forwarded will be held in the mail box for each mobile terminal These terminals can query the hub for down loading messages in their mail boxes
- There will be a distress priority for distress messages originating from any mobile terminal
- Reporting service from mobile terminals to PSTN data links and telex network System will ensure an error free message transfer to HUB

1 4 2 Space Segment

The INSAT MSS space segment consists of two transponders viz a forward link Transponder (c×s) and a return link Transponder (s×c) The details of space segment and the MSS payload is given in [1]

The forward link transponder receives the uplink signals from the HUB station in the 6450 MHz to 6470 MHz band translates, amplifies and transmits them in the 2500 MHz to 2520 MHz band to the mobile terminals The return link transponder receives the uplink signal from the mobile terminals in the 2670 MHz to 2690 MHz band and translates them to the 3680 MHz to 3700 MHz band The forward link channel uses a full bandwidth of 20 MHz, while the return link channel has two slots of 9 MHz each viz 3680 MHz 3689 MHz and 3691 MHz 3700 MHz, one of which will be selected for operation in a given satellite

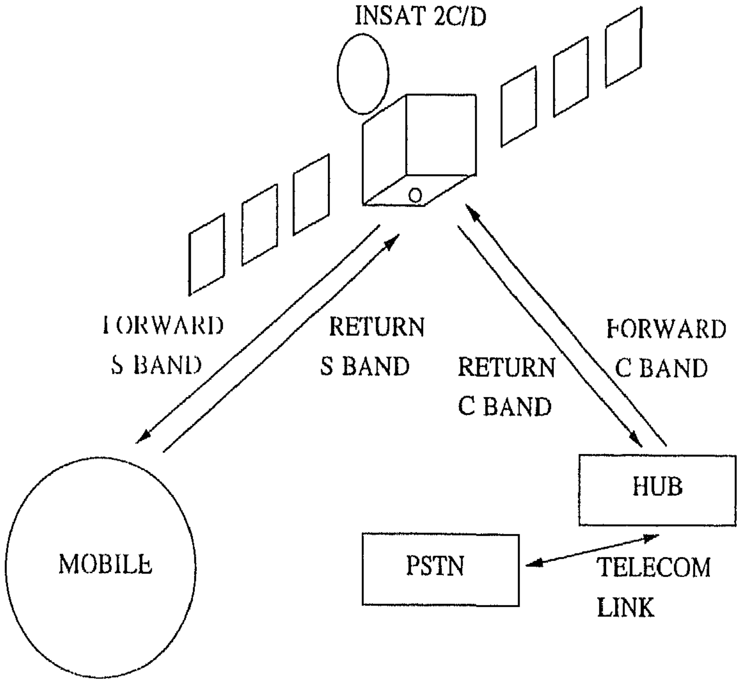


Figure 1 2 INSAT MSS network configuration

1 4 3 Mobile Terminals

Mobile terminals are classified based on the services provided and the operating environment as shown below

- Type A1 Voice/data services for land mobile users
- Type A2 Voice/data services for maritime users
- Type A3 Voice/data services Portable terminals
- Type B1 Messaging service for land mobile users
- Type B2 Messaging service for maritime users
- Type B3 Messaging service Portable terminals
- Type C Reporting service

1.4.4 The HUB Station

HUB provides the central control for the MSS system and acts as the Network Management Center. It provides gateway connectivity to PSTN and telex networks. It will also transmit Bulletin Board information on Time Division Multiplexed (TDM) channels which will contain network information pertaining to channel assignments, return Slotted ALOHA traffic etc.

The HUB station will be equipped with appropriate communications and signaling channel facilities. It will also have appropriate access control, message switching/routing processors etc. All the INSAT MSS services listed in section 1.4.1 will be supported by the HUB. These include providing connectivity between mobile terminals (Type A) and PSTN subscribers for voice, data and fax transmissions. It will store all telex messages originating from Type B terminals and forward it to telex subscribers and will provide a positive acknowledgement on successful telex transmissions. Similarly in coming telex messages will be forwarded to mobile users. It will also prioritize distress messages, acknowledgements. HUB will also act as Mail box server so that mobile users and registered PSTN subscribers can call and collect their messages from the HUB. It will receive and recover error free 'Reporting' messages from Type C terminals and forward them to PSTN and telex subscribers.

1.4.5 Signaling and Channel Access

INSAT MSS uses single channel per carrier (SCPC) channel on demand assignment for Class A type of services. A TDM signaling channel is used for channel assignment purpose by the HUB. The return signaling by the mobile terminals is done by Slotted ALOHA access.

Messaging services for Class B terminals use a TDM message channel in the forward direction from the HUB. The return messaging from the mobile terminals uses Slotted ALOHA access.

Reporting services by Class C terminals use a simple ALOHA access. Different types of INSAT MSS channels are shown in Fig. 1.3 and are summarized in Table 1.1.

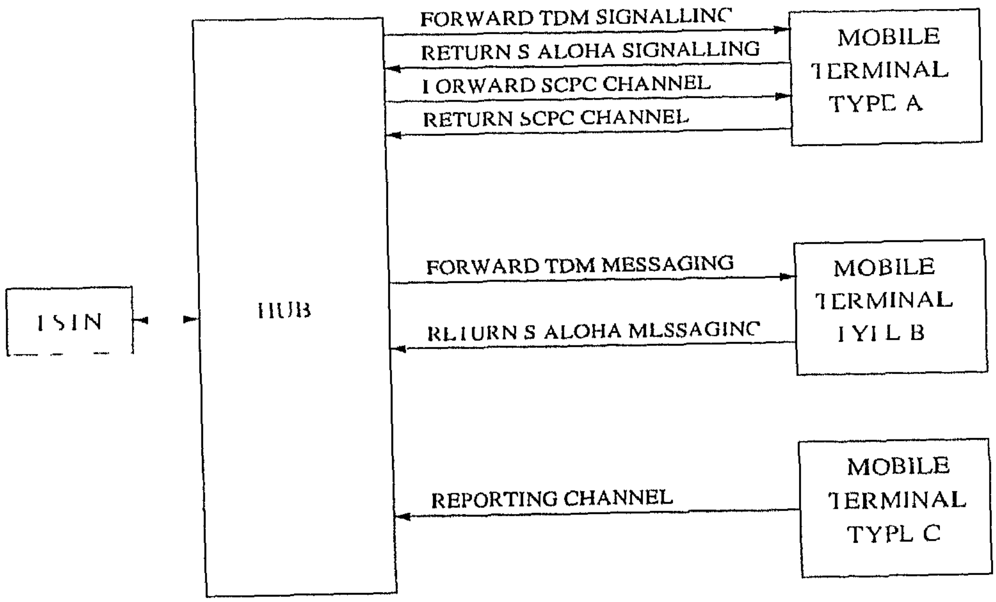


Figure 1 3 INSAT MSS channel configuration

1 5 Organization of Thesis

The rest of the thesis is organized as follows

In chapter 2 a brief review of simulation procedures and the algorithm for simulation methodology used is presented
The protocols and procedures used for Class A type of service and implementation details of its simulator are given in chapter 3 and chapter 4 respectively
Further in chapter 4 the various performance characteristics for this system along with its analysis based on these are presented
Subsequent chapters contains details about Class B type of system
In particular, chapter 5 describes the details of this system
The simulator details are given in chapter 6
A detailed analysis of various performance characteristics along with the drawbacks highlighted from these are also presented in chapter 6
In chapter 7 we present some conclusions along with suggested modifications for Class-B type of system

CHANNEL / SERVICE	LINK	CHANNEL RATE bps	MODULATION METHOD
SCPC (VOICE)	HUB TO MOBILE and MOBILE TO HUB	8000	BPSK
SCPC (DATA)	HUB TO MOBILE and MOBILE TO HUB	8000	BPSK
TDM SIGNALLING	HUB TO MOBILE	1200	BPSK
S ALOHA SIGNALLING	MOBILE TO HUB	1200	BPSK
TDM MESSAGING	HUB TO MOBILE	1200	BPSK
S ALOHA MESSAGING	MOBILE TO HUB	1200	BPSK
REPORTING	MOBILE TO HUB	400	PM

Table 1.1 INSAT MSS communication and signalling channels

Chapter 2

SIMULATION

2.1 Introduction

To study a system we first describe it in terms of a model. This model can then be simulated for the purpose of analysis. We must then look for particular tools to do this. In this chapter we describe simulation techniques and then concentrate on the "Discrete time event" type of simulation, which has been used here to simulate the system.

2.2 Simulation Models

Simulation models are usually classified along three different dimensions.

- Static Vs Dynamic Simulation Models. A static simulation model is a representation of a system at a particular time, or one that may be used to represent a system in which time plays no role; examples of static simulations are Monte Carlo models [6]. On the other hand, a dynamic simulation model represents a system as it evolves over time, such as a queuing system.

- Deterministic Vs Stochastic Simulation Models. If a simulation model does not contain any probabilistic components, it is called deterministic; a complicated system of differential equations describing a chemical reaction might be such a

model. In deterministic models the output is determined once the set of input quantities and relationships in the model have been specified. Many systems, however, must be modeled as having at least some random input components, and these give rise to stochastic simulation models. Most queuing and inventory systems are modeled stochastically. Stochastic simulation models produce output that is itself random, and must therefore be treated as only an estimate of the true characteristics of the model.

- Continuous Vs. Discrete Simulation Models: We define discrete simulation as one for which the state variables change instantaneously at separated points in time. A continuous simulation concerns the modeling over time of a system by a representation in which the state variables change continuously with respect to time. Typically, continuous simulation models involve differential equations that give relationships for the rates of change of the state variables with time.

2.3 Discrete-event Simulation

Discrete event simulation concerns the modeling of a system as it evolves over time by a representation in which the state variables change instantaneously at separate points in time. These points in time are the ones at which an event occurs, where an event is defined as an instantaneous occurrence that may change the state of the system.

Because of the dynamic nature of discrete event simulation models, we must keep track of the current value of simulated time as the simulation proceeds, and we also need a mechanism to advance simulated time from one value to another. We call the variable in the simulation model that gives the current value of simulated time as the simulation clock.

Mainly, two principle approaches have been suggested for advancing the simulation clock: next event time advance and fixed increment time advance. Since we have used next event time advance approach for our discrete event simulation model we shall discuss it here.

With next event time advance approach the simulation clock is initialized to zero and the times of occurrence of future events are determined The simulation clock is then advanced to the time of occurrence of the most imminent of these future events at which point the state of the system is updated to account for the fact that an event has occurred and our knowledge of the times of occurrence of future events is also updated Then the simulation clock is advanced to the time of the new event, the state of the system is updated, and future event times are determined etc This process of advancing the simulation clock from one event time to other is continued until eventually some prespecified stopping condition is satisfied Since all state changes occur only at event times for a discrete event simulation model periods of inactivity are skipped over by jumping the clock from event time to event time Successive jumps of the simulation clock are generally variable in size

The major components of a discrete event simulation model are

- System state The collection of state variables necessary to to describe the system at a particular time
- Simulation clock A variable giving the current value of simulated time
- Event list A list containing the next time when each type of event will occur
- Statistical counters Variables used for storing statistical information about the system performance
- Initialization routine A subprogram to initialize the simulation model at time zero
- Timing routine A subprogram that determines the next event from the event list and then advances the simulation clock to the time when that event is to occur
- Event routine A subprogram that updates the system state when a particular type of event occurs (there is one event routine for each event type)
- Library routines A set of subprograms used to generate random observations from probability distributions that are a part of the simulation model

- Report generator: A subprogram that computes estimates (from the statistical counters) of the desired measures of performance and produces a report when the simulation ends.

- Main program: A subprogram that invokes the timing routine to determine the next event and then transfers control to the corresponding event routine to update the system state appropriately. The main program also checks for termination and invokes the report generator when the simulation is over.

The logical relationships (flow of control) among these components is shown in Fig. 2.1.

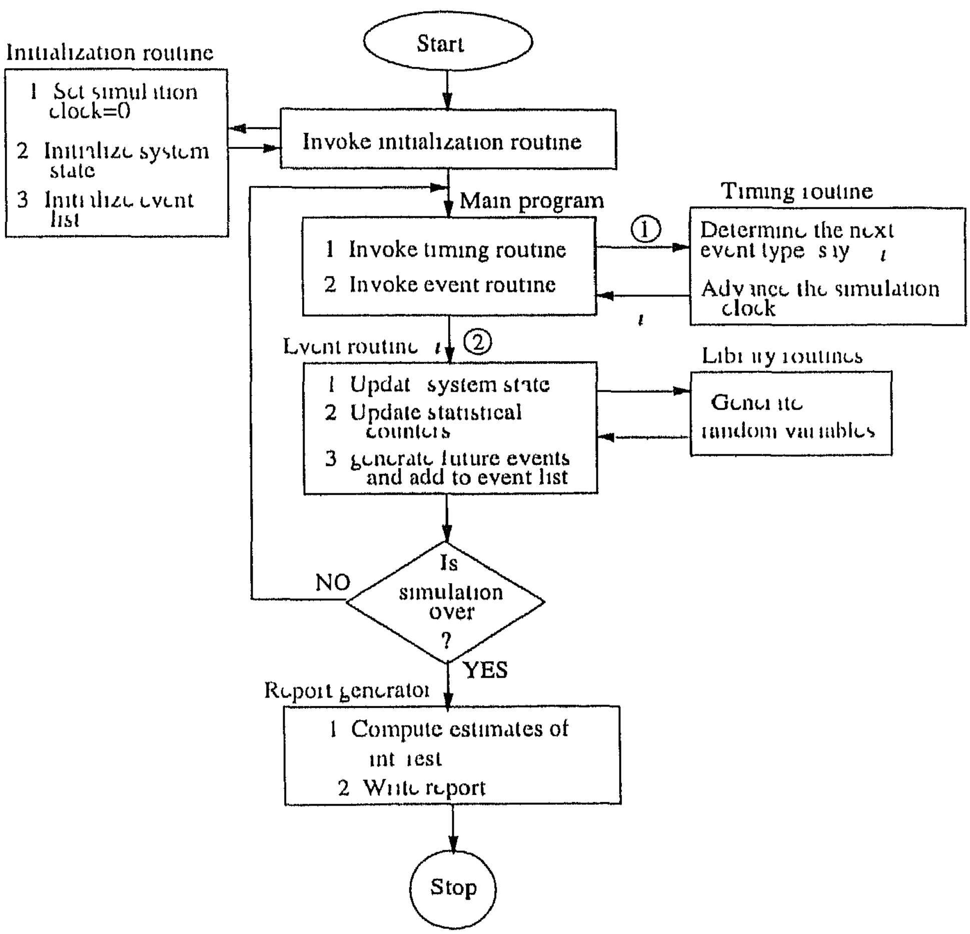


Figure 2.1 Flow of control for the next event time advance approach

2.4 Simulation of a Single-Server Queue

The simulation carried out in this thesis work is based on single server queuing models. We have used such multiple queues to represent the systems of interest. We now describe the simulation of the M/D/1 [5] type of queue which has served as the basic element of our model. Fig 2.2 contains a flowchart for the arrival event. When an arrival events occurs, the time of the next arrival in the future is generated and placed in the event list. Then a check is made to determine whether the server is busy. If so the number of customers in the queue is incremented by one, and we ask whether the storage space allocated to hold the queue is already full. If the queue is already full, an error message is produced and the simulation is stopped, if there is still room in the queue, the identifier (pointer) pointing to the data structure associated with that arriving customer is put at the end of the queue. On the other hand if the arriving customer finds the server idle, then this customer has a zero delay. The server must be made busy, and the time of departure from service of the arriving customer is scheduled into the event list.

The departure event's logic is depicted in the flow chart of Fig 2.3. This routine is invoked when a service completion occurs. If the departing customer leaves no other customer behind in queue, the server is idled and the departure event is eliminated from consideration, since the next event must be an arrival. On the other hand if one or more customers are left behind by the departing customer, the first customer in queue will leave the queue and enter service so the queue length is reduced by one, and a departure event for the customer now entering service is scheduled. Finally, the rest of the queue (if any) is advanced one place. The simulation will begin in the "empty and idle" state, i.e. no customers are present and the server is idle. At time 0, we will begin waiting for the arrival of the first customer, which will occur after the first interarrival time, rather than at time 0. Our simulation shall terminate after a fixed number of customers have been served by the system as we shall see in the following chapters. The time at which the simulation ends is thus a random variable.

In order to get an account of various performance measures we make use of some counters which are to be constantly updated over the period of simulation , some

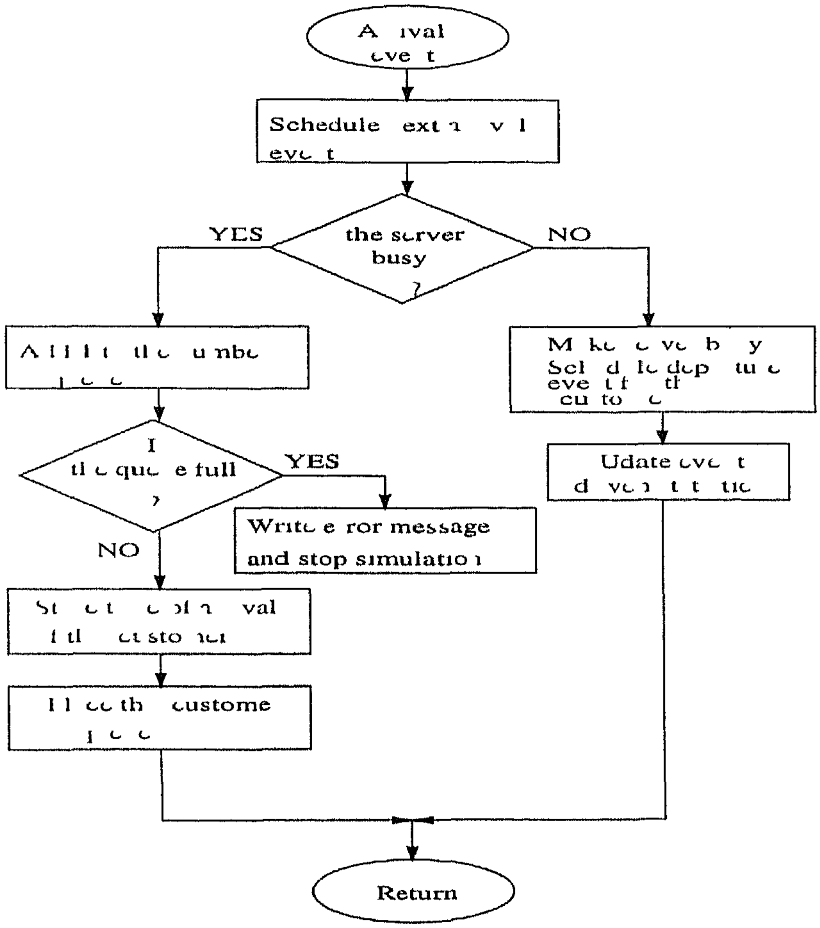


Figure 2 2 Flowchart for arrival routine, queuing model

of these counters will be updated as and when some event will occur while some other counters have to be updated continuously as the simulation time proceeds For updating such counters we jump to the routine to update these statistical counters after every event, one such measure is the expected queue length, denoted by q To compute this let $Q(t)$ denote the number of customers in queue at time t for any real number $t \geq 0$, and let T be the time over which we have simulated Then for any time t between 0 and T, $Q(t)$ is a nonnegative integer, and if we let p_i to be expected proportion of the time that $Q(t)$ is equal to i, then definition of q will be

$$q = \sum_{i=0}^{\infty} i p_i \tag{2 1}$$

Thus, q is a weighted average of the possible values i for the queue length $Q(t)$, with the weights being expected proportion of time the queue spends at each of its possible lengths To estimate q from a simulation, we replace the p_i's with estimates

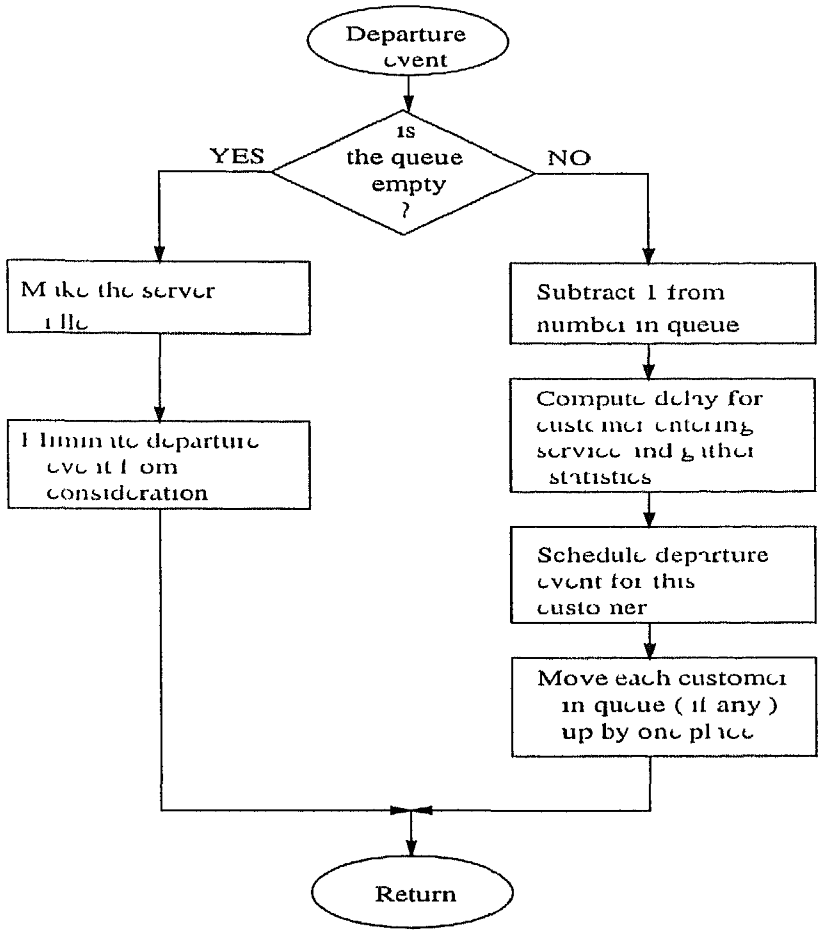


Figure 2 3 Flow chart for departure routine, queueing model

of them, and get

$$q = \sum_{i=0}^{\infty} i p_i \tag{2 2}$$

where p_i is the observed proportion of the time during the simulation that there were i customers in the queue Computationally , it is easier to rewrite q using some geometric considerations If we let T_i be the total time during the simulation that the queue is of length i then $T = T_0 + T_1 + T_2 +$ and $p_i = T_i/T$, so that we can rewrite Eq 2 2 above as

$$q = \frac{\sum_{i=0}^{\infty} i T_i}{T} \tag{2 3}$$

This equation can be realised in the simulation by finding the difference between the time of the previous event and the time at the current event At every event

instance such time slices are found and then the sum of product of these time slices and the number in the queue for these time durations is found to get the sum in the numerator of Eq 2 3, the estimate of the mean can be then found by dividing this sum by the total time duration T

To simulate the model we need a way to generate random variables with uniform and exponential distributions The uniform random numbers can be generated by invoking a random number generator to generate a variate U that is distributed uniformly between 0 and 1 this distribution is referred to as $U(0,1)$ and has a probability density function

$$f(x) = \begin{cases} 1 & \text{if } 0 \leq x \leq 1 \\ 0 & \text{otherwise} \end{cases}$$

Now to obtain a exponential random variate we shall take the natural logarithm of it, multiply the result by β the mean, and finally change the sign, that is $-\beta \ln U$ For our purpose we have made use of the library routine drand48(), to generate uniform random numbers This function returns random number between 0 and 1 with unform distribution It makes use of linear congruential equation and 48 bit arithmetic, and returns a nonnegative, double precision, floating point number uniformly distributed over the range $0 < y < 1$ 0

2 5 Confidence Intervals and Hypothesis Test for The Mean

Since simulation models use random variables as input, the simulation output data are themselves random and care must be taken in drawing conclusions about the model's true characteristics e g the average queue length described above Therefore in order to validate the model and obtain a reasonable estimate of the output means, we performed confidence interval tests [6] for our simulation model

If x_1 $x_2, x_3,$, x_n are normal random variables, then $100(1-\alpha)$ percent confidence interval for the mean μ is given by

$$\bar{x}(n) \pm t_{n-1,1-\alpha/2}\sqrt{\frac{S^2(n)}{n}}$$

where $t_{n-1,1-\alpha/2}$ is the upper $1-\alpha/2$ critical point for the t distribution [6]. These critical points are given in [6].

$$\bar{x}(n) = \text{Sample Mean} = \frac{\sum_{i=1}^{n} x_i}{n}$$

$$S^2(n) = \text{Sample Variance} = \frac{\sum_{i=1}^{n}[x_i - \bar{x}(n)]^2}{n-1}$$

Chapter 3

CLASS-A SERVICES PROTOCOLS AND PROCEDURES

3 1 Introduction

The Class A type of service will provide communication for the voice, data and Fax using single channel per carrier (SCPC) communication channels on demand assignment basis The HUB acts as the gateway between the terrestrial and mobile net works It also acts as a network management center and provides for the centralized control The SCPC channels will be held in a pool at the HUB The HUB performs the function of assignment of these channels based on their demand and availability The signaling for Class A type of services will be In band, Sub band and Out of-band The Out of band signaling is performed using the Time Division Multiplexed (TDM Assignment) channel from the HUB to mobile terminals A Slotted ALOHA (Request) channel will be used by the mobile terminals for the Out of band signaling in reverse direction For the PSTN (Public Switched Telephone Network) originated calls, the HUB transmits the call announcement message in the TDM channel (Out of-band) and mobile terminals respond in the Slotted ALOHA channel SCPC channel assignments will be made by the HUB using the TDM channel The In band and Out of band signaling is done in the SCPC channels The mobiles will have a num bering scheme of six digits as follows

Y Z Z Z Z Z

Y Stands for the service type

1= Telephone 2=FAX 3=DATA

The mobile ID will be a sequence of 24 bits A unique pair of IDs will be allocated for each mobile terminal These two different IDs will be used for the forward and return paths The HUB will maintain a cross reference table relating the mobile IDs with the six digit mobile number The 24 bit ID will be included in all the transmissions from the HUB to a specific mobile and from the mobile terminals to the HUB The details of the numbering scheme are given in [4]

3 2 Signal Unit (SU) Format

The signal units to be be used for the Out band, In band and the Sub band signaling messages are identical in their structure The signal units will carry the different types of messages as listed in Table 3 1

Each signal unit will be of 12 bytes The first byte will always identify the message type and the last two bytes (11th and 12th) will carry the CRC code The specific details for each signal unit and their different fields are given in [4] Some of the signals can be of more than one SU For these signaling messages, the first SU is the "Initial signal unit (ISU) This is then followed by the "subsequent signal unit" (SSU) Bulletin Board (BB) messages will be transmitted in the TDM channel These messages provide the information regarding the signaling channel frequencies and network status to the mobile terminals

3 3 Channels

Two types of channels will be used in the system

SIGNAL TYPE (HEX)	FUNCTION TYPE	SOURCE	MODE / CHANNEL
8 1	Call Announcement	HUB	TDM
8 2	Mobile Response	Mobile	S ALOHA
8 3	Access Request	Mobile	S ALOHA
8 5	Call Failure Indication	HUB/Mobile	TDM/SA
8 6	Channel Assignment	HUB	TDM
8 8	Service Address (ISU)	Mobile	In band
8 9	Service Address (SSU)	Mobile	In band
8 A	Channel Release	HUB/Mobile	In band
8 B	Return Carrier ID	Mobile	In band
8 C	Connect	HUB	Sub band
8 D	Scrambling Vector	Mobile	In band
9 0	Bulletin Board (BB)	HUB	TDM
9 1	Bulletin Board (BB)	HUB	TDM
9 2	Fill in SU	HUB	Sub band
9 4	Selective Clear	HUB	TDM In band
9 5	Scrambling Vector ACK	HUB	In band
9 6	Group ID Update	HUB	TDM
9 7	Group ID ACK	Mobile	S ALOHA
9 9	Mobile Connect	Mobile	Sub band
9 A	Maintaining Burst Indicator	HUB	In band
9 C	Request Ch Randomization (BB)	HUB	TDM
9 D	Group Call Announcement	HUB	TDM
9 E	Group Call Ch Assignment	HUB	TDM
A 0	Group Call Clear	HUB	In band
A 9	Group Call Failure Indication	HUB	In band

Table 3 1 Signal Types For CLASS A Service

- Communication channels
- Signaling channels

3.3.1 TDM signaling (Assignment) channel

The TDM assignment channel will operate at 1.2 Kbps and is expected to have a Bit Error Rate (BER) of at most 1×10^{-6}. Each TDM frame will be of 1.32 seconds. It will carry seven signal units of 96 bits each and a BB packet of 96 bits. These signal units are rate 1/2 FEC coded. A 32 bit Unique Word (UW) is then appended to give the TDM frame of 1584 bits as seen in Fig. 3.1.

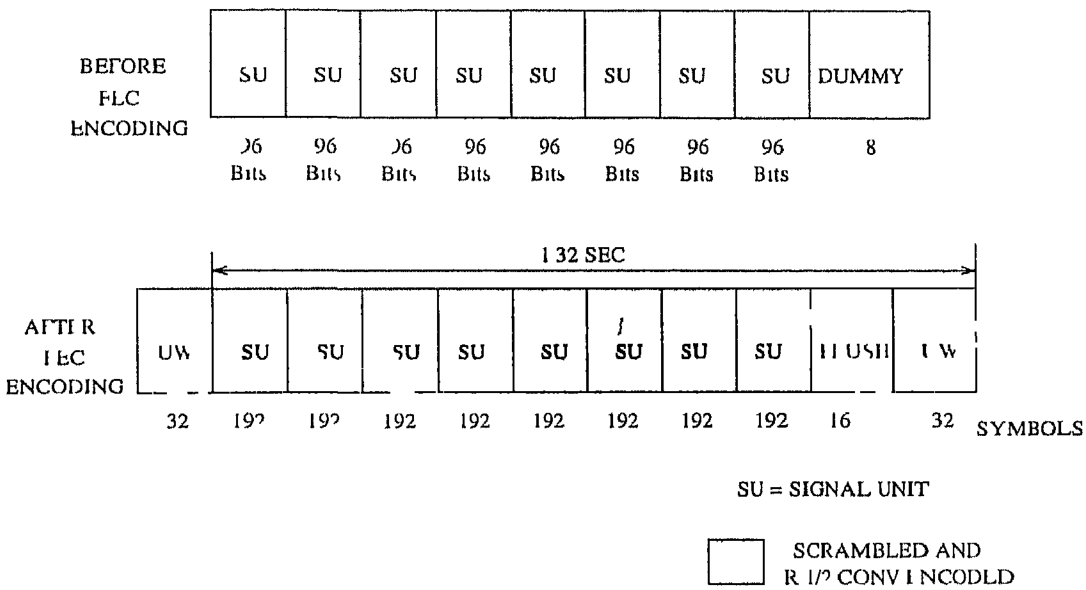


Figure 3.1 TDM channel frame format

3.3.2 Slotted ALOHA signaling (Request) channel

There is a single Slotted ALOHA channel for carrying the Out of band SU from the mobiles to the HUB. This channel will operate at 1.2 Kbps and is expected to have a BER of at most 1×10^{-6}. During each TDM frame of 1.32 sec, there will be 4 burst slots. These burst transmissions will be synchronized to the incoming TDM frame slots as shown in Fig. 3.2. Each burst will carry a single signal unit of 12 bytes. This

signal unit is rate 1/2 FEC coded Each burst will consist of a 20 bit preamble (CW), a Unique Word (UW) pattern the SU and 16 flush bits to flush out the encoder The burst format for Slotted ALOHA channel is shown in Fig 3 2 coding and other details for it are given in [4]

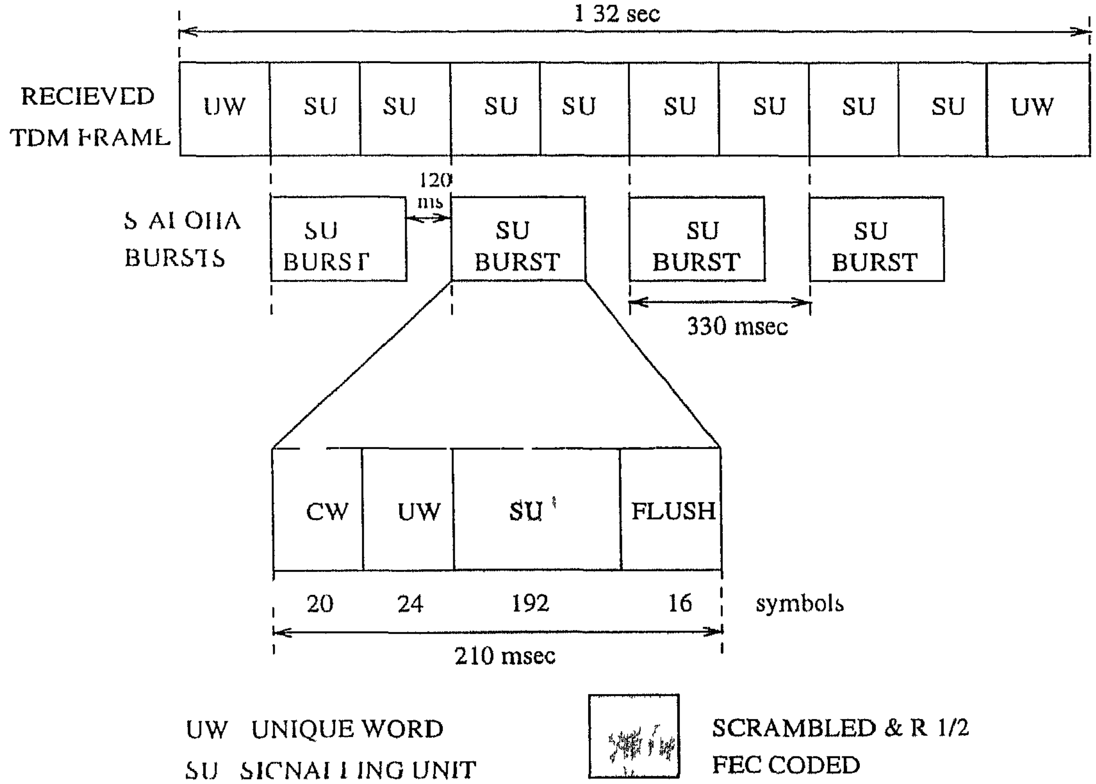


Figure 3 2 Slotted Aloha burst format

3 3 3 SCPC communication channels

These channels are held in a pool at the HUB, which assigns them on demand for communication between the HUB and the mobile terminals The SCPC channels will operate in three modes viz ,"Voice", **"Data"** and **"In band signaling"**

Voice-mode The frame structure for SCPC channels operating in this mode is given in Fig 3 3

Each burst of voice transmission will be divided into 240 ms voice frames Each 240 ms voice frame will be subdivided into 4 subframes of 60 ms each The voice

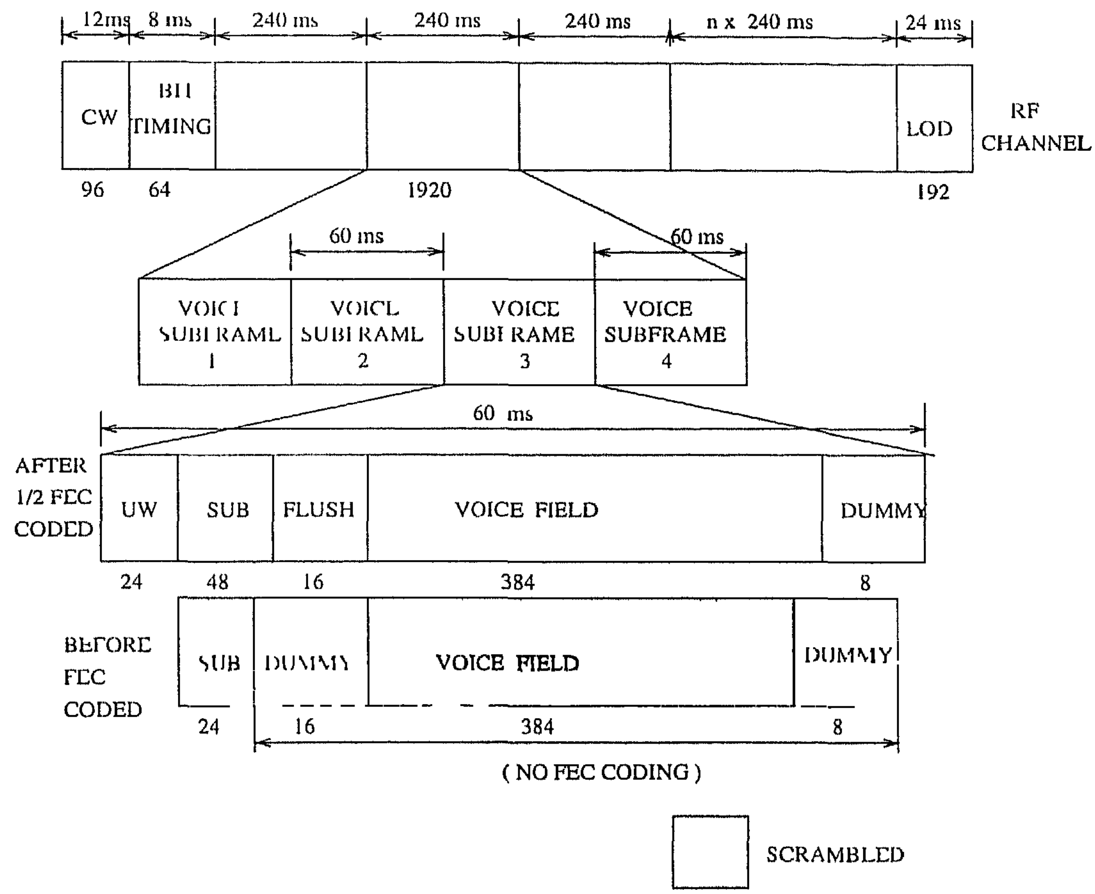


Figure 3 3 SCPC channel frame format in voice mode

subframe will carry 384 bits of voice information and 24 bits of Sub band signal unit The 24 bits of Sub band signal units are rate 1/2 FEC coded, no FEC coding is done for the information bits A standard SU of 96 bits is obtained by combining the Sub band bits from all the four subframes in a 240 ms voice frame One signal unit will be transmitted as a Sub band signal during a single 240 ms voice frame The coding and other details are given in [4]

Data mode In the data mode the rate of data transfer will be 2400 bps in the continuos mode In this mode, the frame structure will be similar to the voice mode Rate 1/2 FEC coding will be applied here to both the data and Sub band fields The information field in the data sub frame will be filled with 288 bits of data block There will be four such subframes during each 240 ms data frame Each data frame will be repeated after one sub-frame to provide adequate redundancy in order to protect the transmissions against short duration fades

in the signal However 24 Sub band signaling bits will not be repeated The data mode configuration of the SCPC channel is shown in Fig 3 4

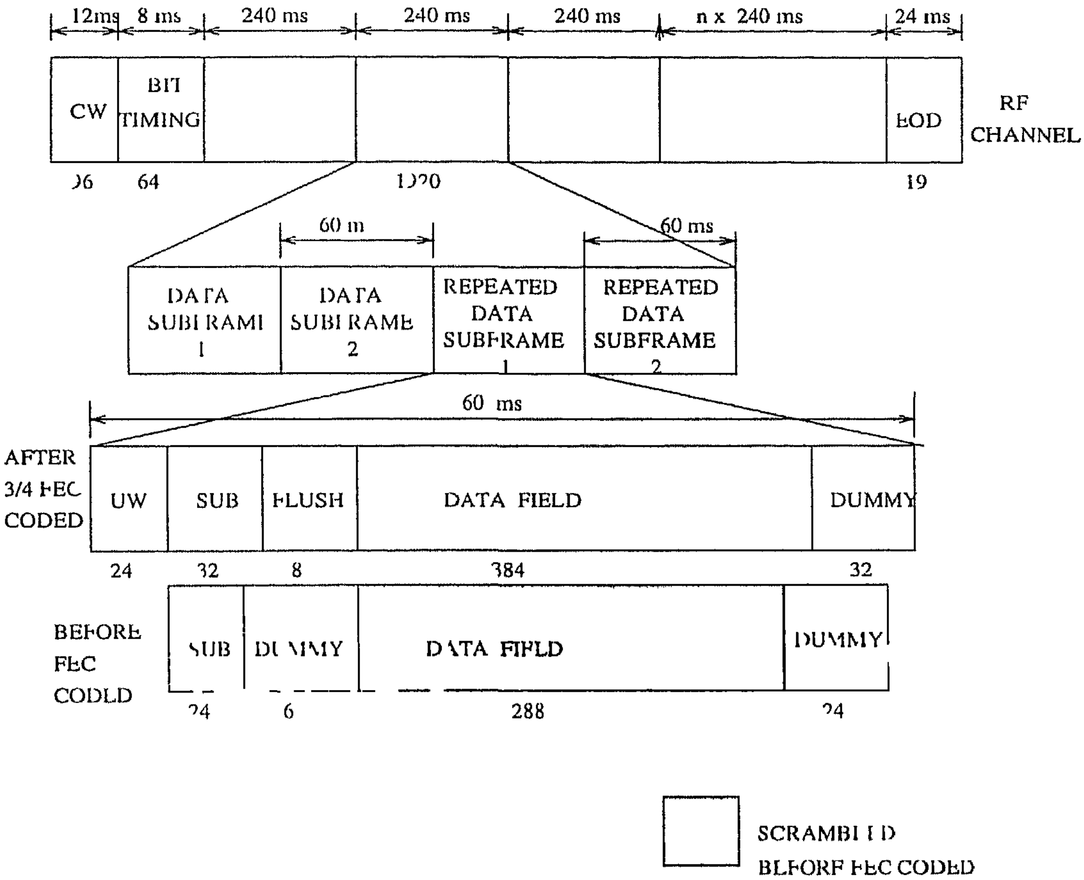


Figure 3 4 SCPC channel frame format in data mode

In-band signaling mode The In band signaling mode also uses a frame format similar to the ones for the 'voice" and 'data' mode Each 240 ms frame is divided into four subframes Each subframe will consist of 2 copies of 96 bits In band SU a 24 bit Sub band field, a UW pattern and the flush bits The In band SU will be repeated in all the 4 subframes Thus the same SU will be transmitted eight times during one subframe of 240 ms This frame structure is shown in Fig 3 5

The communication in a SCPC channel starts with the channel being in the In-band mode It then switches to the "voice" mode or "data" mode as required Finally, for terminating the call, the SCPC channel switches back to the 'In band" mode

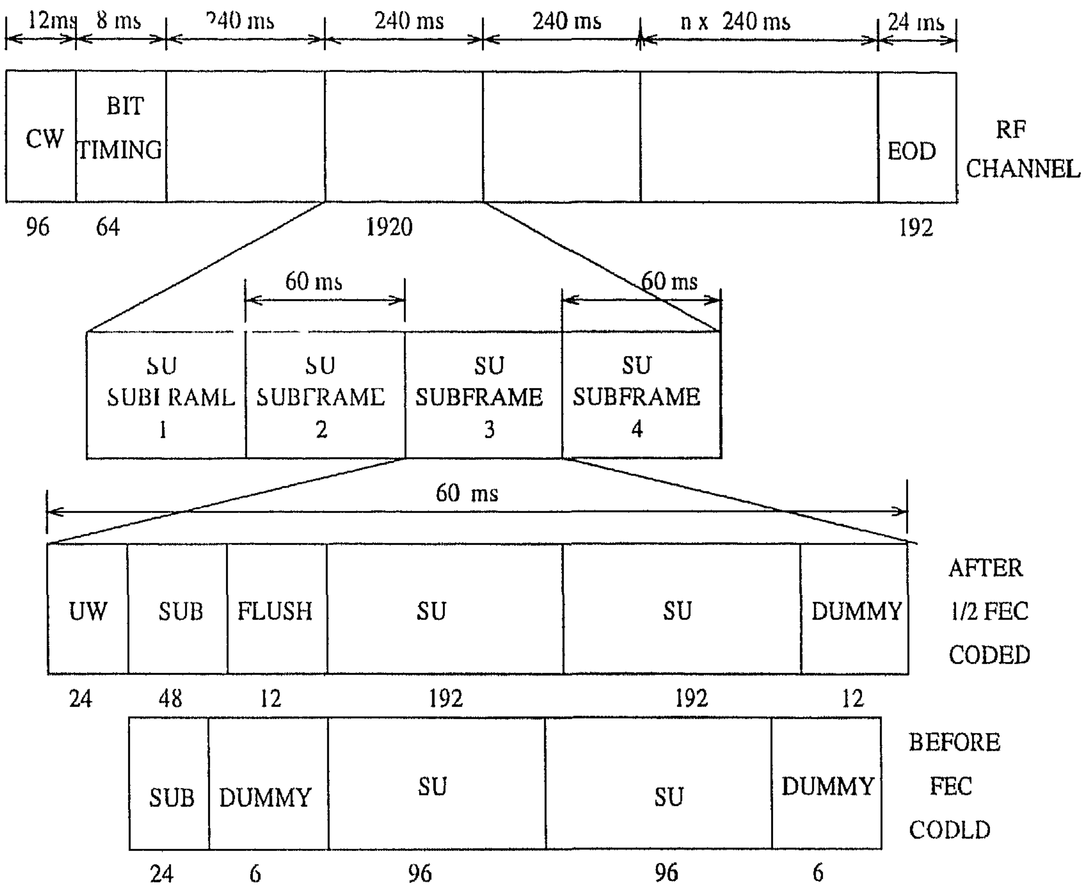


Figure 3 5 SCPC channel frame format in In band mode

3 4 Mobile Originated Call Set-up Procedure

The mobile terminal starts the call set up by sending a "Access Request signal (83H) to the HUB in the Slotted ALOHA channel The HUB on receiving a access request checks for the following

- Status of mobile terminal (The call can be set up only if the mobile terminal is not in the BUSY list")
- If that service unit is available
- If the mobile is authorized for that service
- If the SCPC channels are available

If all these conditions are satisfied the HUB assigns two SCPC channels to the mobile and sends the "Channel Assignment" (86H) signal to the mobile terminal in the TDM channel The HUB also places the mobile in the BUSY list and updates the SCPC channel record The channel assignment is sent 3 times with a separation of 2 TDM subframes (0 5 sec) This redundancy is provided to ensure successful reception of signals at the mobile terminal If any of the above mentioned conditions is not satisfied the HUB transmits a "Call Failure Indication" (85H) in the TDM channel After transmission of the "Channel Assignment" message the HUB starts In band signaling and waits for a time duration (TH1) for obtaining the "Scrambling Vector" from the mobile If no response is received within this time period, the HUB starts the call clearing sequence

If the response from the HUB is not received within a specified time duration (TM1), the mobile retransmits the 'Access Request' signal The mobile will retry this signal for a maximum of three times, if no response is obtained, the mobile will indicate a call failure and return to the idle state

On receiving the channel assignment, the mobile terminal tunes to the assigned SCPC channels The mobile terminal then sends the "Service Address" (88H and 89H) and the "Scrambling Vector" (8DH) as In band signal units in the SCPC channel These signals will be sent continuously by the mobile until it receives the "Scrambling Vector ACK" (95H) from the HUB, or a timer (TM2) expires On correct reception of "Scrambling Vector ACK" from the HUB, the mobile sends the "Return Carrier ID" (8BH) continuously until the transmission of ACK is stopped by the HUB or a timer (TM3) expires The mobile transits to the "voice" mode when the transmission of ACK from the HUB is stopped

The HUB on receiving the "Scrambling Vector" from the mobile transmits the "Scrambling Vector ACK" continuously and starts a timer (TH2) The "Scrambling Vector ACK" will be transmitted until it receives the " Return Carrier ID", from the mobile or the timer expires

The HUB starts the PSTN call set up procedure on receiving the "Return Carrier ID" (8BH) from the mobile terminal The sequence of signaling to be followed for

the mobile originated call set up is summarized in Fig 3 6

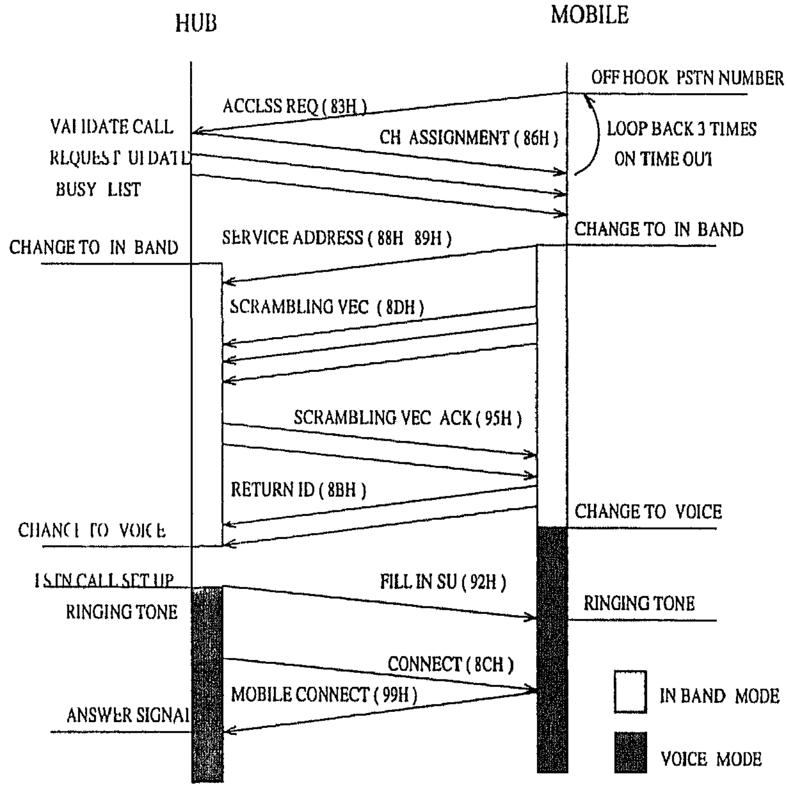


Figure 3 6 Mobile originated call set up procedure

3 5 PSTN Originated Call Set-up

The HUB receives the mobile number from the PSTN subscriber It then checks for the presence of mobile in the "BUSY list" If the mobile is not marked "BUSY" the HUB checks for the availability of the requested service, mobile authorization and the availability of SCPC channels prior to the call set up If all the above mentioned conditions are satisfied the HUB proceeds with the call establishment with mobile unit It sends a 'Call Announcement" (81H) signal to the mobile in the TDM channel and places the mobile in the "BUSY list" The HUB then waits for a certain time duration (TH20) to receive the Mobile Response" (82H) It retransmits "Call Announcement' if the response fails to arrive before the timeout Maximum number of such retransmissions will be three before the call is declared to have failed

The mobile on receiving a "Call Announcement" will transmit the "Mobile Re

sponse" (82H) in the Slotted ALOHA channel The mobile then waits for a certain time duration (TM20) to receive the "Channel Assignment' (86H) It retransmits the response message if the channel assignment signal fails to reach it before the timeout Maximum number of such retransmissions will be three

On receiving the 'Response' signal from the mobile, the HUB transmits the "Channel Assignment" signal three times with a separation interval of two TDM subframes (0 5 sec) The HUB then switches to In band mode and waits for a certain time duration (TH21) for receiving the "Scrambling Vector" from the mobile

The mobile terminal will tune to the assigned SCPC frequencies on receiving the 'Channel Assignment' message and starts In band signaling It then transmits 'Scrambling Vector" (8DH) continuously This signal will be transmitted continuously until it receives a Scrambling Vector ACK" (95H) or a timer (TM21) expires On receiving the ACK the mobile transmits the "Return Carrier ID" (8BH) and generates the "ring" signal for the mobile telephone interface This signal will be sent continuously until the HUB stops transmission of the ACK or a timer (TM22) expires When the HUB stops the transmission of ACK, the mobile switches to the voice" mode and proceeds with the call by sending the "Connect" signal to the HUB

The HUB on receiving the "Scrambling Vector" transmits the "Scrambling Vector ACK" (95H) continuously until it receives the "Return Carrier ID" or a timer (TH22) expires On receiving the "Return Carrier ID" the HUB stops the transmission of "Scrambling Vector ACK" and sends ring back to the PSTN subscriber It then switches to "voice" mode and waits for receiving the "Mobile Connect' signal The sequence of signaling to be followed for the PSTN originated call set up is summerized in Fig 3 7

On the expiry of timers at the HUB or at mobile, the call clearing procedures are invoked if they are transmitting in the In band voice or data mode The call clearing procedure is similar to that invoked for call clearing on the normal termination of calls This is described in Section 3 6

The procedures and signaling followed for the set-up of data calls will be same as for the voice calls, in the Out of band and In band signaling modes In case of

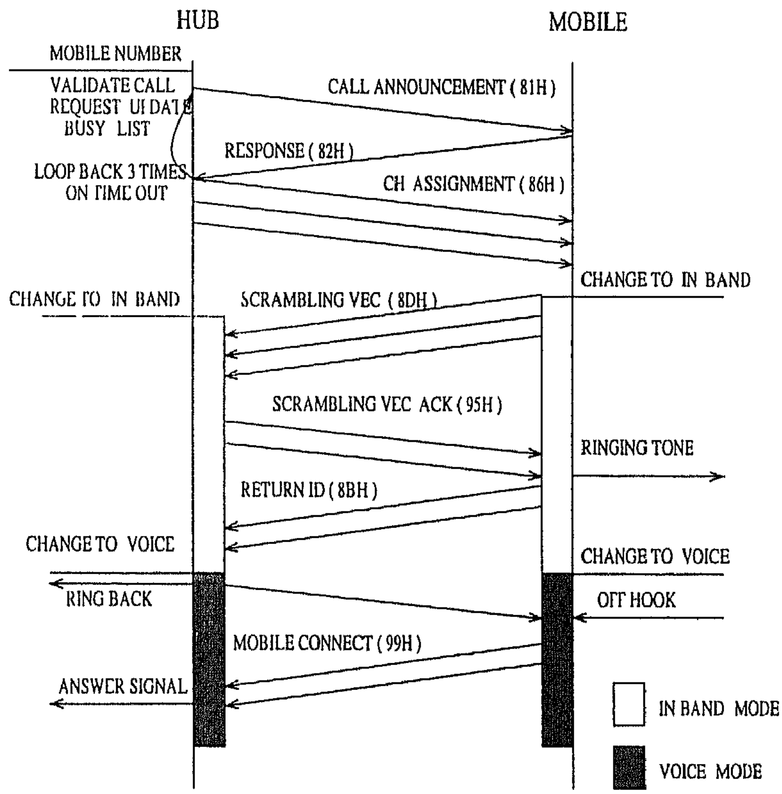


Figure 3 7 PSTN originated call set procedure

a data call the HUB and the mobile terminal switches to the 'data' mode after successful completion of In band signaling The signaling in the 'data" mode is as per V 22bis procedures The details for these are given in [4] After the call is established no simulation is carried out for servicing it in the "voice" or 'data" mode It is assumed that after successful completion of In band signaling the call proceeds until it terminates

3 6 Clearing Of Mobile Originated Calls

When the call clearing sequence is invoked, the side initiating it transits to the In band signaling mode Fig 3 8 shows the signaling sequence for HUB and mobile initiated clearing of calls

3 6 1 Mobile Initiated Clearing

The mobile terminal initiates the clearing sequence by first transiting to the In band signaling mode It sends six Channel Release" (8AH) messages with appropriate "Cause Indication' to the HUB The mobile then turns off it's carrier

The HUB on receiving the Channel Release" message turns it's own carrier off, if the mobile carrier has been turned off The HUB also updates the list of available SCPC channels and removes the mobile from the "BUSY list" If the mobile carrier is not found to be off within certain time duration, the HUB initiates the call clearing sequence followed for the HUB initiated call clearing'

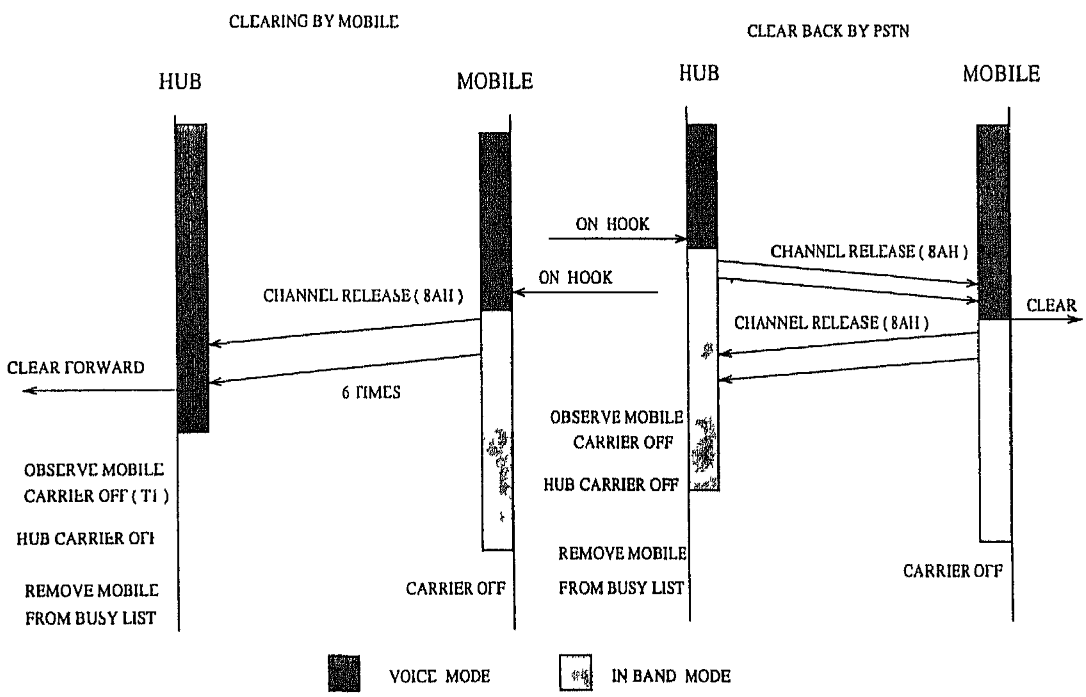


Figure 3 8 HUB and mobile initiated call clearing

3 6 2 HUB Initiated Clearing

The call clearing sequence from the HUB is initiated by switching to the In-band mode It then sends Channel Release" (8AH) signal continuously to the mobile terminal with appropriate cause indication value [4] The HUB stops the transmission of "Channel Release" (8AH), only when it receives the "Channel Release' signal from

the mobile terminal or if a timer expires On receiving the 'Channel Release from the mobile terminal the HUB removes the mobile from the BUSY list and updates the list of available SCPC channels The HUB waits for a certain time duration for the mobile carrier to be OFF If the carrier is not found to be OFF after this time duration, appropriate action is taken [4]

The simulator is designed for the calls to be cleared normally It does not consider the event for which the mobiles fail to switch the carrier OFF, and hence the HUB may have to initiate the call clearing sequence for a malfunctioning mobile unit Also the simulator does not considers the case, where the mobile carrier is found to be OFF by the HUB without receiving the "Channel Release' from the mobile

Chapter 4

CLASS-A. SIMULATION MODEL AND ANALYSIS

4.1 Simulation Model

The Class A type of system described in the previous chapter can be modeled as shown in Fig 4.1. Using this model a discrete time event type of simulator of this system has been developed. This has been used to study the performance of system and find typical values for it's operating parameters. We developed the discrete time event type of simulator using C. The functional blocks of the model have been emulated in the simulator by employing various data structures. The various event routines control the operation of the system as it evolves over time and implements the functions that the system performs during it's operation. The major components of the system and the data structures used for simulating them in software are summarized below. In the simulator the effect due to channel fading is neglected.

4.1.1 Functional Blocks

A basic data structure has been defined that contains all the major fields to identify and process the important signaling units described in the previous chapter. This also provides an aid in emulating the communication of certain call characteristics between the HUB and the mobile. This data structure is illustrated in Fig 4.2. We call this

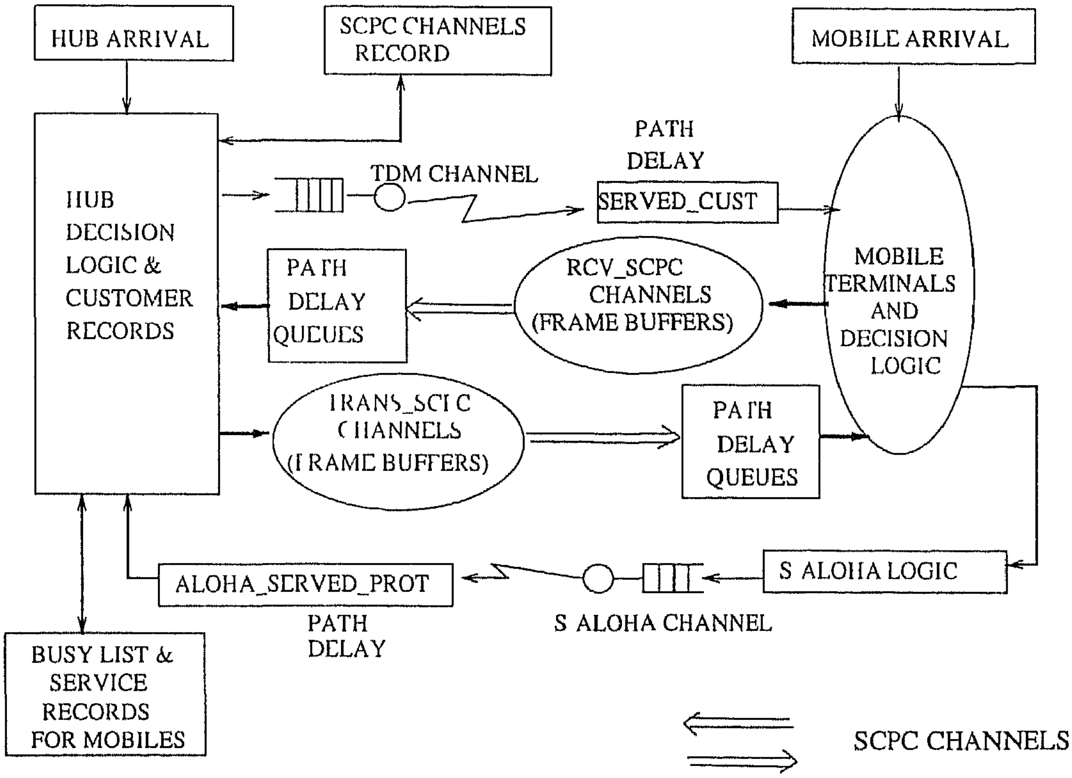


Figure 4 1 Model for Class A type of system

as cust The pointers to objects of this class are passed through queues, representing the channels for communication between the HUB and the mobile terminals These pointers may also be encapsulated within the data structure of type frame, defined to identify and process a frame within the SCPC channels in voice data or In band mode

The data structure of type frame is illustrated in Fig 4 3 The pointers to objects of this class are passed through the queues representing SCPC channels for communication between the HUB and mobile terminals

HUB The data structures representing the components of HUB are

customer This is the representative list of customers at the HUB, elements of the array customer shall contain the pointer to particular object of type cust This object has certain details of the call and the associated signals

mob_cust_busy This is the list of status, of mobile terminals maintained at the

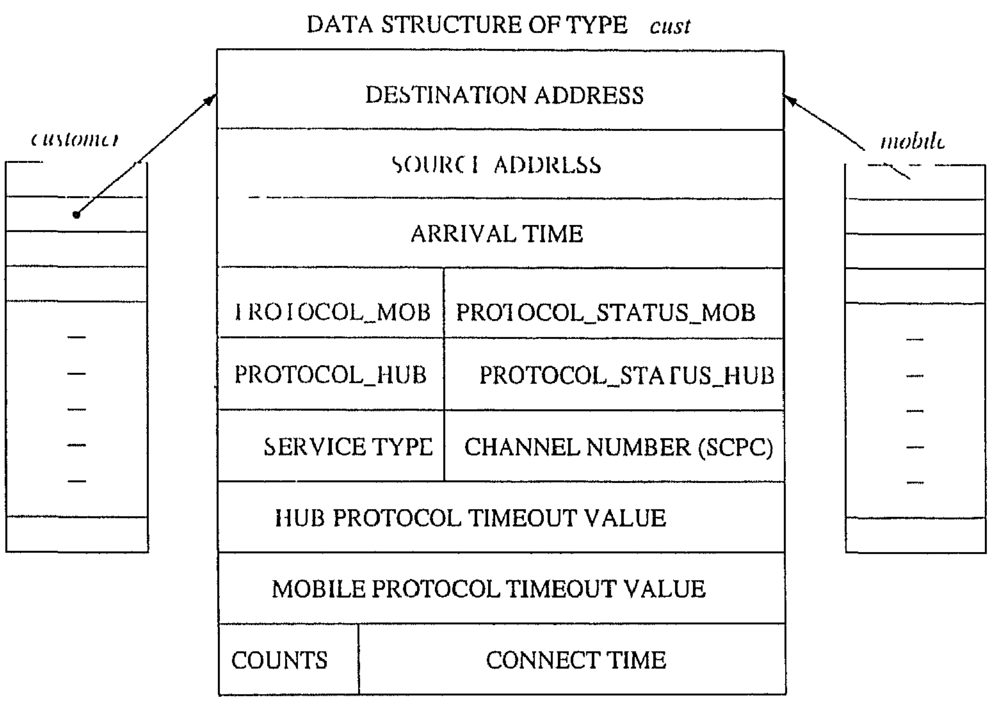


Figure 4.2 The data structure cust

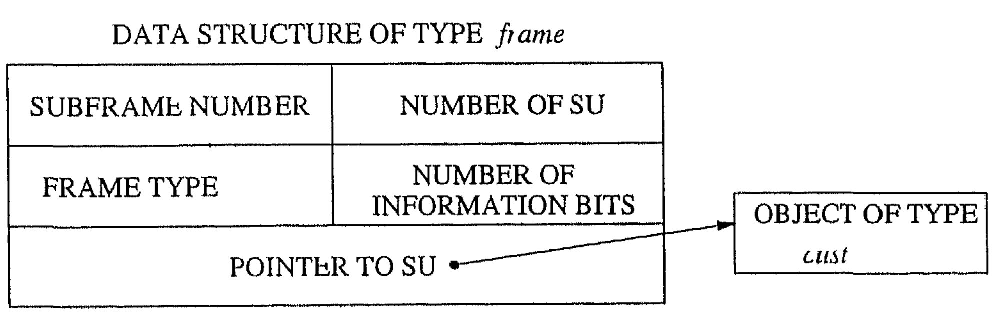


Figure 4.3 The data structure frame

HUB These can be BUSY or FREE depending on whether the corresponding mobile terminal is busy or free

customer_list This list contains the status of each customer at the HUB This also stores some associated parameters necessary for processing the call The parameters stored are

Customer status The status can be one of the following six types

- Voice call initiator
- Voice call called
- Data call initiator
- Data call called
- Fax call initiator
- Fax call called

Connection duration The duration for which the call is requested

Channel number The SCPC channel assigned for communication between the HUB and mobile

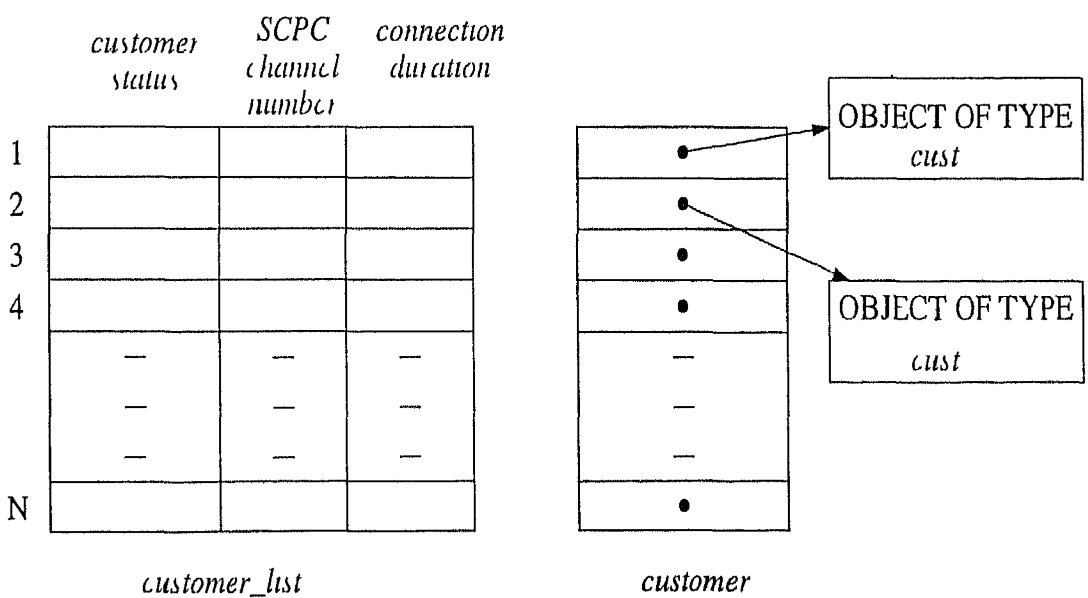


Figure 4 4 Data structures for representing the customers at HUB

Time division multiplexed (TDM) Channel The data structures used for representing this channel are

queue1 This data structure maintains the list of SU to be transmitted in TDM channel. It emulates the queue for TDM channel. Each element can contain pointer to an object of type cust.

server1_status Maintains the status of server for the TDM queue at the HUB.

served_cust This array contains pointers to signal unit serviced from the TDM channel. This is used to simulate the queue necessary to introduce the path delay in the TDM channel. The path delay of 480 msec for these signaling units is added to the value of time at which the signaling unit is served from the HUB. The value obtained thus is the time instant at which this signaling unit is expected to arrive at the mobile terminal. This value is stored in the array served_cust delay_time.

Mobile Terminals The data structures used for representing them are

mobile This is the representative list of mobile customers. Elements of array mobile will contain the pointers to particular object of type cust. This object has certain details of the call and the associated signals.

mob_list This list contains the status of each mobile customer. It also stores some associated parameters necessary for processing the calls. These parameters are same as those stored in the data structure customer_list.

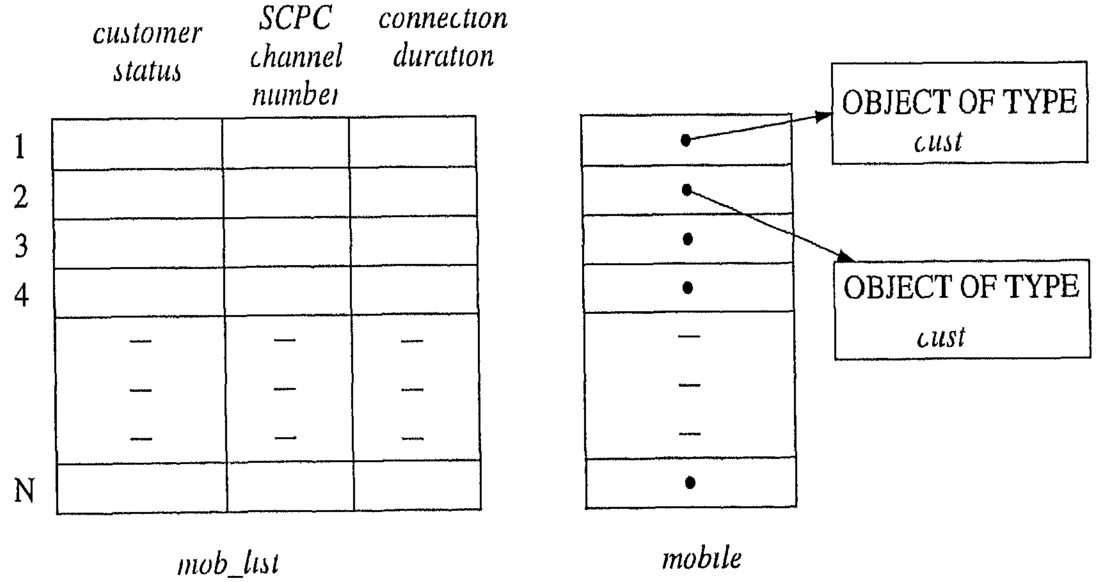


Figure 4.5 Data structures for representing the mobile customers

Slotted ALOHA Channel The Slotted ALOHA channel is simulated using two arrays

pointers This contains the list of signaling units to be served by the Slotted ALOHA channel

transmit_time This is the list of corresponding time when the mobile units are expected to attempt transmission of respective signal units

The Slotted ALOHA is implemented by looking up the transmit_time table at every slot beginning, if there are more than one entry having values less than current value of time corresponding action is taken for the collision condition If only one entry is present then it is served by entering the pointer value in aloha_served_prot Then the pointers and transmit_time tables are up dated The path delay is introduced in a manner similar to that described in for the TDM channel

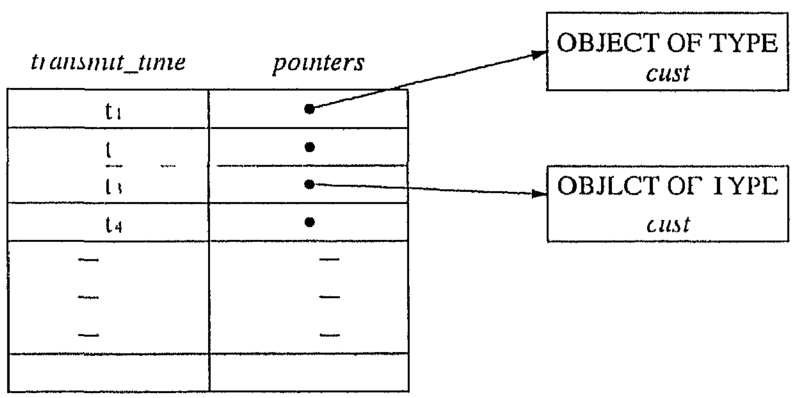


Figure 4 6 Arrays for simulating the Slotted ALOHA channel

Single Channel per Carrier (SCPC) Channels In the simulator the SCPC channels are assumed to be organized in pairs, although the draft[1] states them to be used in an unpaired manner This enables management of SCPC channels to be simplified Each pair consists of one SCPC channel for transmitting from the HUB to mobile This is notified as the trans_channel The other channel is for communication from the mobile to the HUB and is notified as the rcv_channel

[1]The INSAT MSS draft by ISRO

The call when established will have one pair of SCPC channels associated for the communication between the HUB and mobile unit

The SCPC channels pairs and their associated management functions are simulated using the following data structures

trans_frame_buff and rcv frame_buff are the arrays containing pointers to objects of type frame frame carries the information to be transmitted in the forth coming subframe of the SCPC channel These pointers are copied into the arrays trans_frame_que and rcv_frame_que after they are served from the HUB in order to introduce the path delay The corresponding time until which the frames are to be delayed is stored in the arrays rcv_frame_que_time and trans_frame_que time respectively The arrays trans_channel_info and rcv_channel_info contains the information related to the status and the mode of operation of the SCPC channels

4 1 2 System Parameters

HUB_MEAN_INTER The mean interarrival time of calls arriving at the HUB

MOB MEAN INTER The mean interarrival time of mobiles requesting for call

The distribution of interarrival time of calls is exponential [5] with the mean represented by parameters given above The service time distribution for the calls is also assumed to be exponential with mean as one of the following parameters

avg_serv_time_voice Average call holding (service) time for voice calls

avg_serv_time_data Average call holding (service) time for data calls

avg_serv_time_Fax Average call holding time for Fax calls

MAX_NOS_CHANNELS The maximum number of channel pairs (SCPC) available for assignment to the mobiles for communication with the HUB

The system parameters include the various timers that control the time out of protocols during message transfers These timers are represented by

- TH1 (86H)
- TH2 (95H)
- TM1 (83H)
- TM2 (8DH)
- TM3 (8BH)
- TH20 (81H)
- TH21 (86H)
- TM20 (82H)

The characters in brackets indicate the signaling units after transmission of which these timers are set The current value of time added to these timer values gives the time to live value for the corresponding signaling units This time to live value is placed in the hub_timeout or mob_timeout field of respective signaling unit depending on weather HUB is transmitting or the mobile unit is transmitting the corresponding signaling unit

4 1 3 Performance Parameters

Parameters that represent the performance of the system in steady state, have been measured by conducting the simulation run for different sets of system parameters The performance parameters that are likely to represent the steady state characteristics of the system are summarized below

Average traffic carried in Erlangs This is calculated as the sum of the fraction of total time period for which each server is busy [7]

Average channel utilization This is calculated as the average traffic carried per SCPC channel pair

Number of calls blocked This is the number of calls blocked due to nonavailability of SCPC channels

Blocking probability This is the probability that a call request finds all SCPC channels busy

Average call set up time This is measured as the time since the 'Access request is sent (for mobile originated call) or call announcement SU is placed in it's queue (for HUB initiated call) , till the successful reception of connect SU

Average call termination time This is measured as the average time that elapses since the channel release SU (8AH) is transmitted from the mobile for mobile initiated termination till the call is terminated at the HUB and the (SCPC) channels are released For HUB initiated call termination, the HUB transmits the channel release signaling unit For this the average call termination time is then measured as the time since the channel release SU is transmitted from the HUB and until the HUB receives the channel release SU in it's response from the mobile and releases the SCPC channels

Throughput of Slotted ALOHA channel

Call rejections Number of call rejections due to time out of various timers are listed in the data files

4 2 Flow Charts

A Broad outline of the logic followed for the simulation is illustrated in the flow charts of Fig 4 7 and Fig 4 8 Fig 4 7 illustrates the main program that invokes the timing routine to determine the next event and then transfers control to the corresponding event routine to update the system state appropriately The main program also checks for termination and invokes the report generator when the simulation is over Fig 4 8 illustrates the various event routines

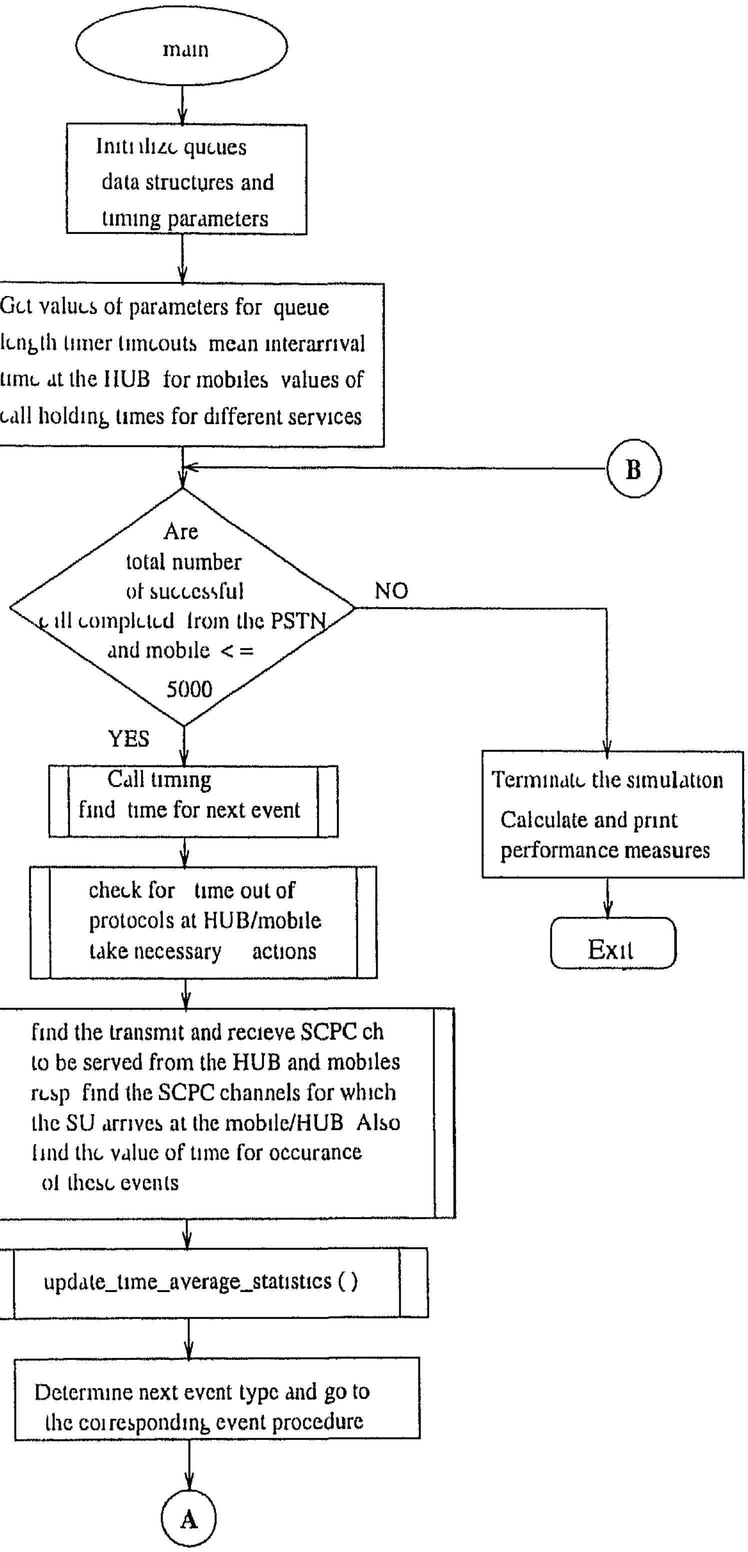


Figure 4 7 Flow chart for the main program

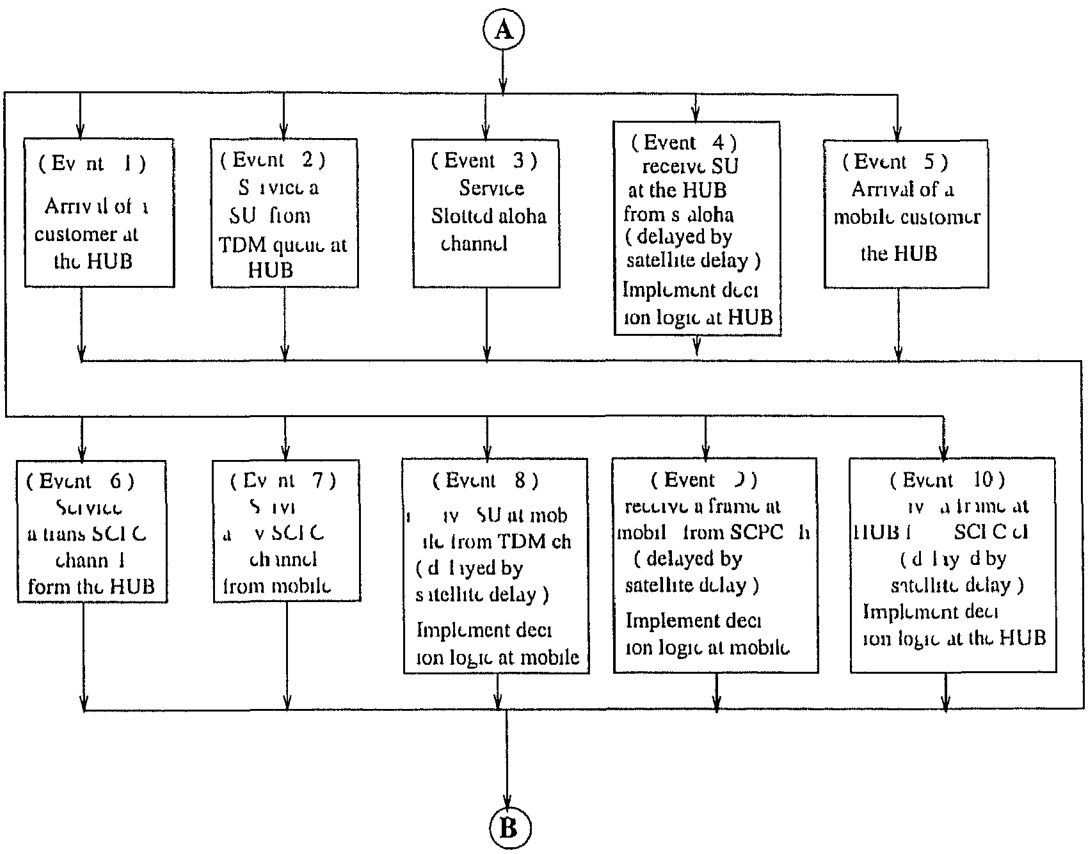


Fig 4 7 continued

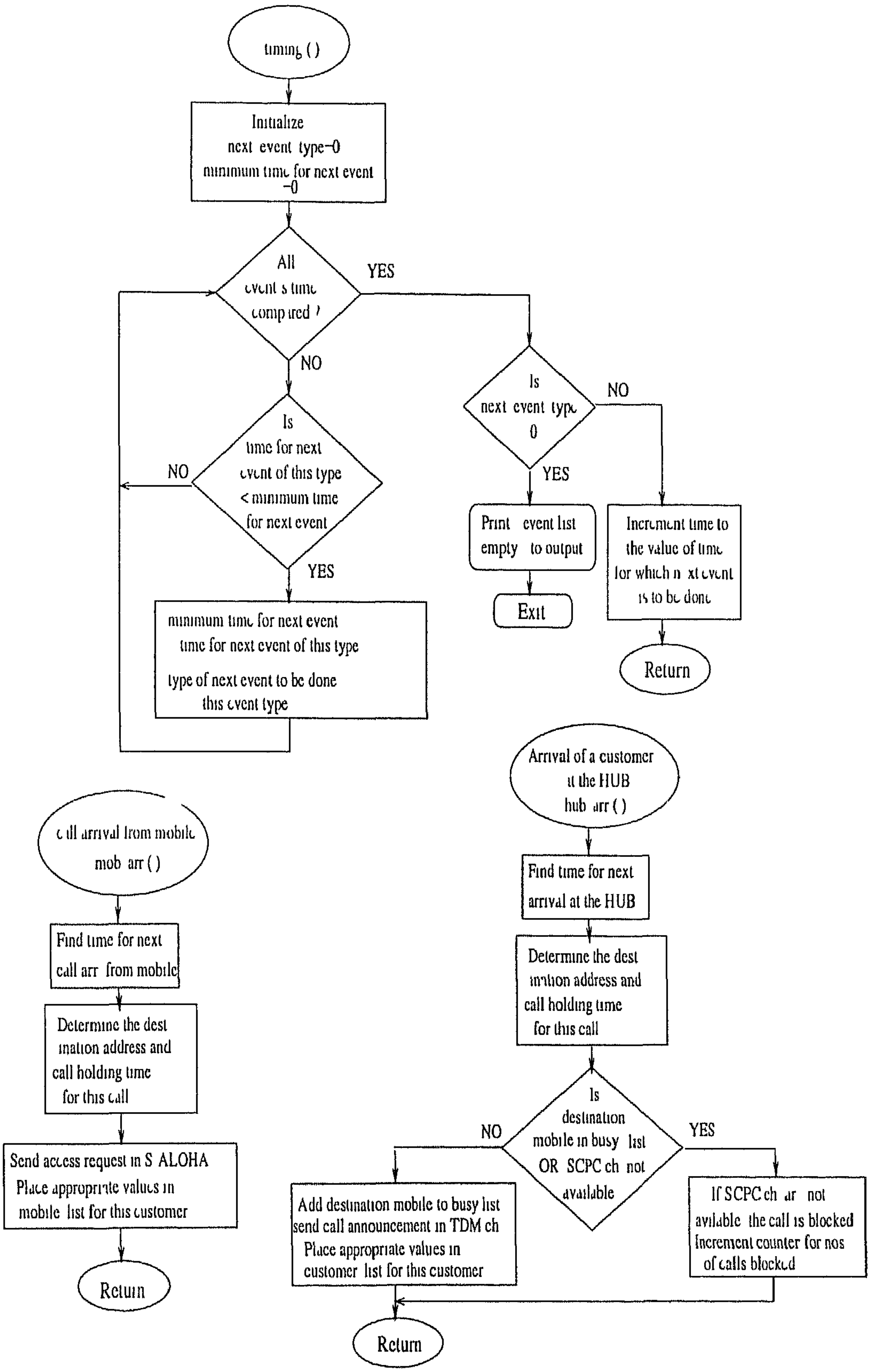


Figure 4 8 Flow charts for the event routines

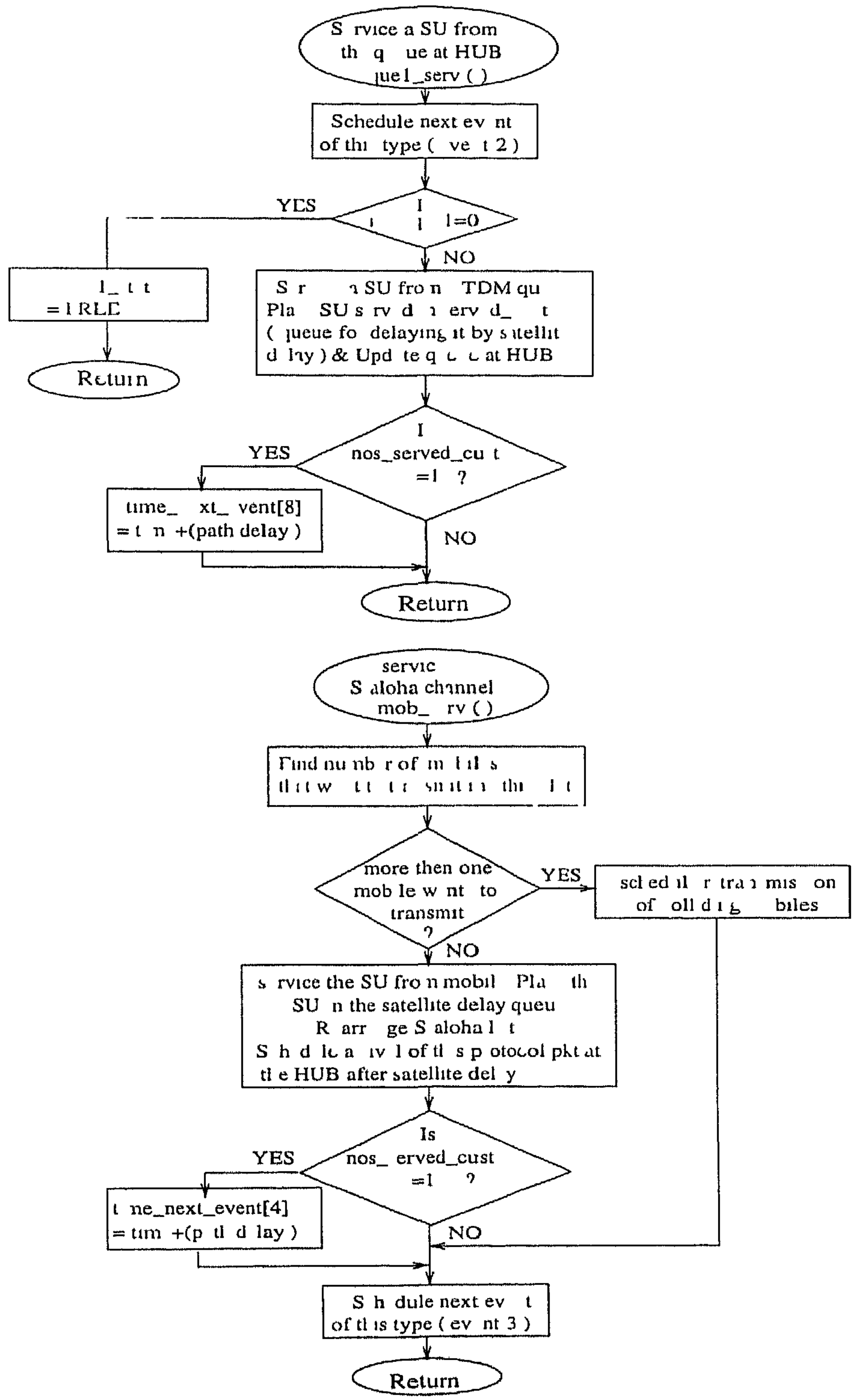
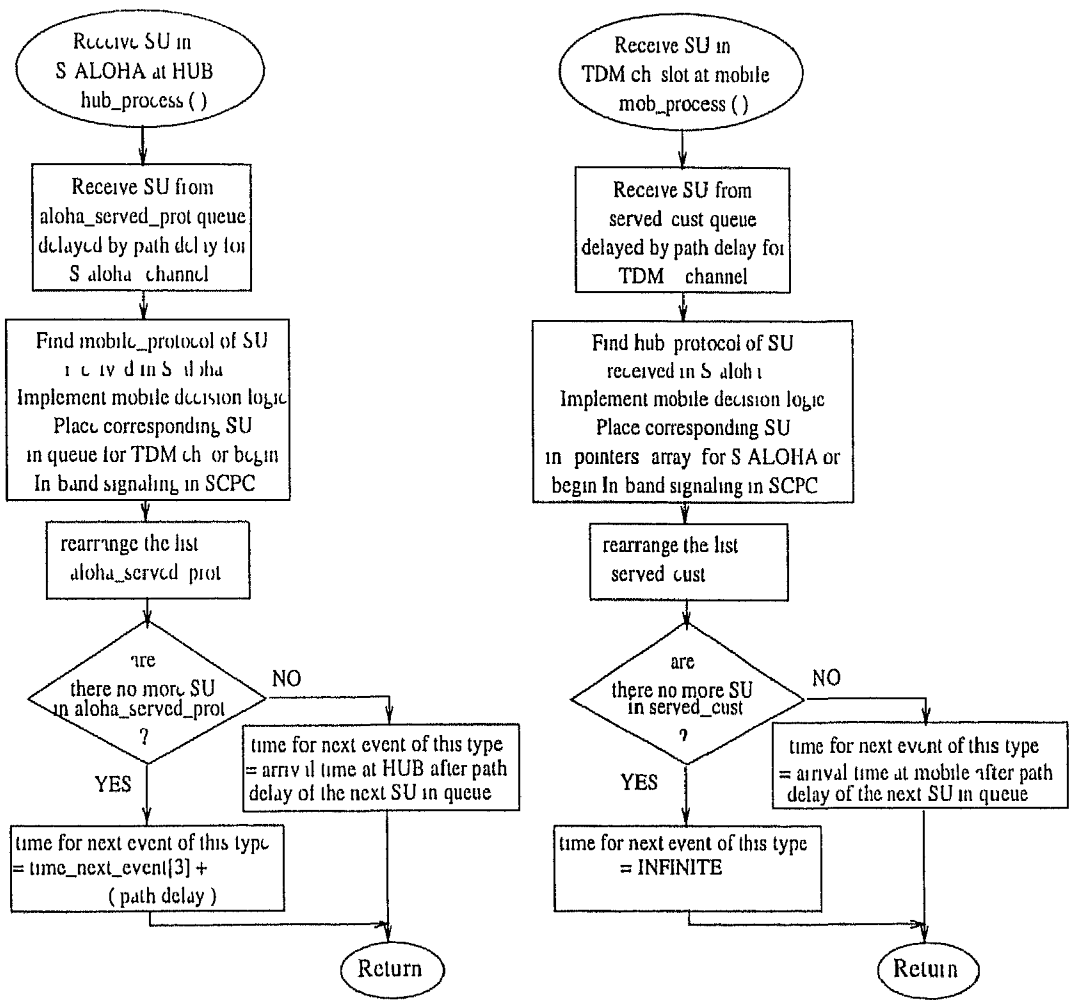


Fig 4 8 continued



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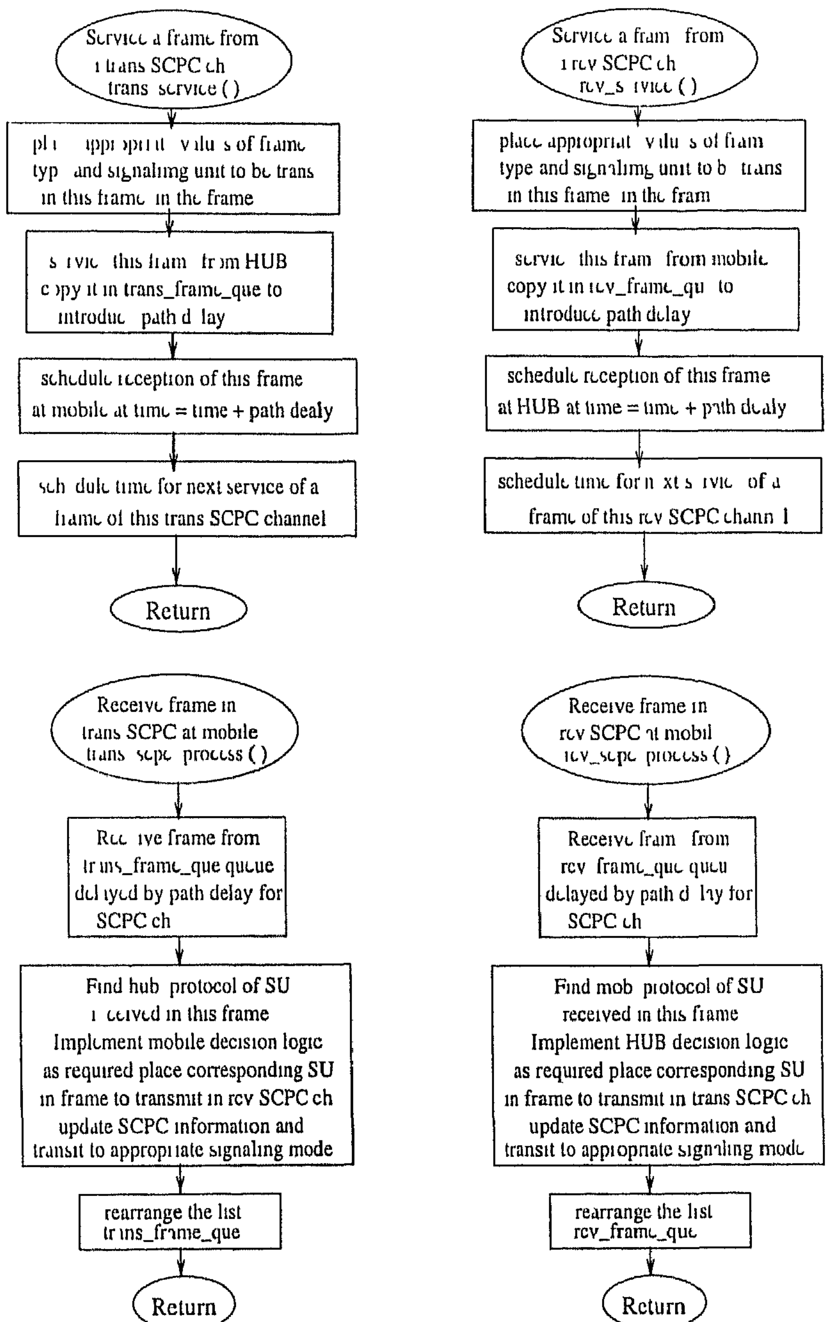


Fig 4 8 continued

4.3 Observations

The Class A type of system for continuous voice and data services is based on 'circuit switching' [7]. The HUB performs the function of assigning the channels i.e. servers for transmitting and receiving to the mobile terminals on requests arriving from the PSTN and mobile subscribers. Although majority of calls in a mobile system are outgoing (60% to 80%) i.e. mobile originated [3], but for the purpose of analysis we have assumed the system to be balanced i.e. equal arrival rates from the PSTN and the mobiles. The traffic arriving to the system is characterized as a Poisson arrival process [5] with exponential interarrival time and call holding time distributions. Identical values of mean interarrival at the HUB and for mobiles are used while experimenting with the system under different traffic condition. This value will hence forth be referred to as the 'mean interarrival time' (MI). The total arrival rate of the calls to the system will hence be 2/MI. The calls can be of three types 'voice', 'data' and 'Fax'. For the simplification of analysis, we assume the call holding times to be the same irrespective of the type of service. Simulations have been carried out for two different values of call holding time parameter: 5 minutes and 3 minutes.

The simulation experiments are carried out assuming an infinite number of customers. This permits us to use Poisson arrival model for traffic characterization. This system can be modeled as a 'Lost Calls Cleared" (LCC) system [7]. Blocking will be observed in the system when all the SCPC channels are busy. In the context of subscriber calls, the LCC model assumes that if the call is blocked the subscriber hangs up and waits for a certain length of time before attempting again. He does not retry immediately or within a short duration of time. Such calls are considered to have been cleared from the system and the reattempts are treated as new calls. According to the LCC model the blocking probability of a call is the probability that all the servers in the system are busy. This blocking probability is obtained using the famous Erlang B formula or the loss formula.

$$P_n = \frac{\frac{A^n}{n!}}{\sum_{i=0}^{n} \frac{A}{i!}}$$

A = Offered traffic in Erlangs

n = Number of servers (channel pairs)

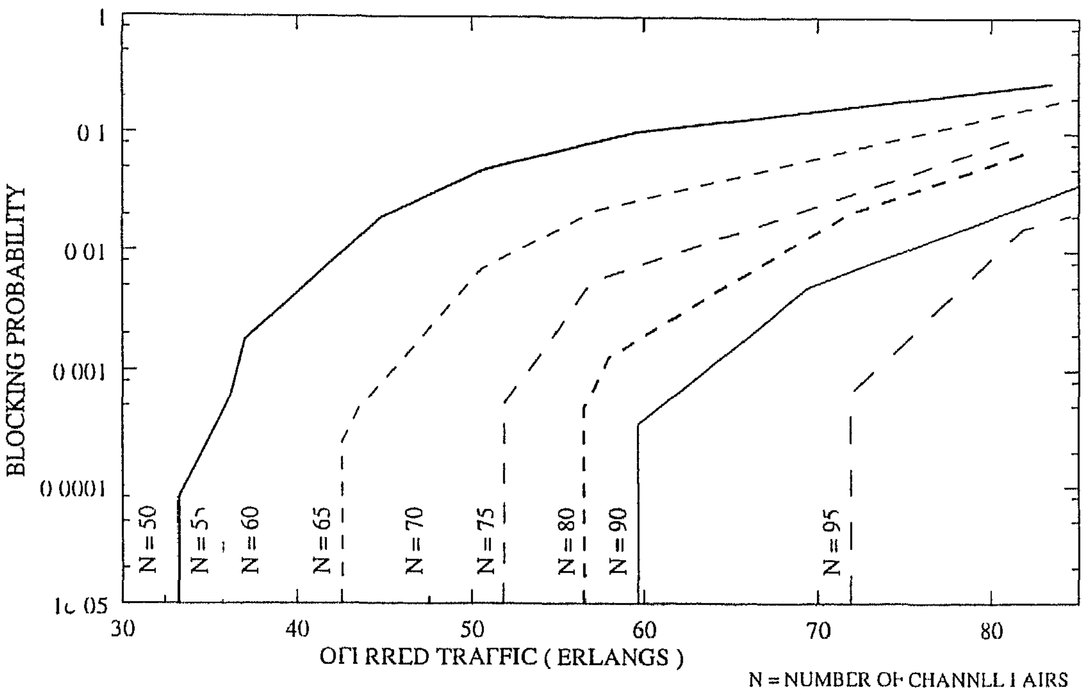


Figure 4.9 Blocking characteristics of the Class A system

The blocking characteristic of the system, obtained from the simulation data is illustrated in Fig 4.9. The nature of plots obtained is found to be similar to that for the Erlang B formula[2]. For a particular value of blocking probability and N (number of channel pairs) the traffic offered to the system is observed to be higher than the corresponding value of traffic listed in the Erlang B tables. This is observed because a small fraction of the traffic offered to the system is lost in the Slotted ALOHA (request) channel and the TDM (assignment) channel, in the process of call establishment. The blocking probability is observed to be extremely low for N greater than 100, for the range of offered traffic illustrated in Fig 4.9.

The blocking characteristic for different values of mean interarrival time and call holding time (AS), and N is illustrated in Fig 4.10. Once again in this figure we observe an extremely low value of blocking probability with N greater than 100, for values of mean interarrival time greater than 6 seconds and call holding time ≤ 5 minutes. It is observed to be maximum for mean interarrival time of 7 seconds and call holding time of 5 minutes. No blocking is observed with N greater than 110 for values of mean interarrival time greater than 6 sec, and call holding time less than or

[2] Erlang B plots and tables are given in the book Digital Telephony by Bellamy

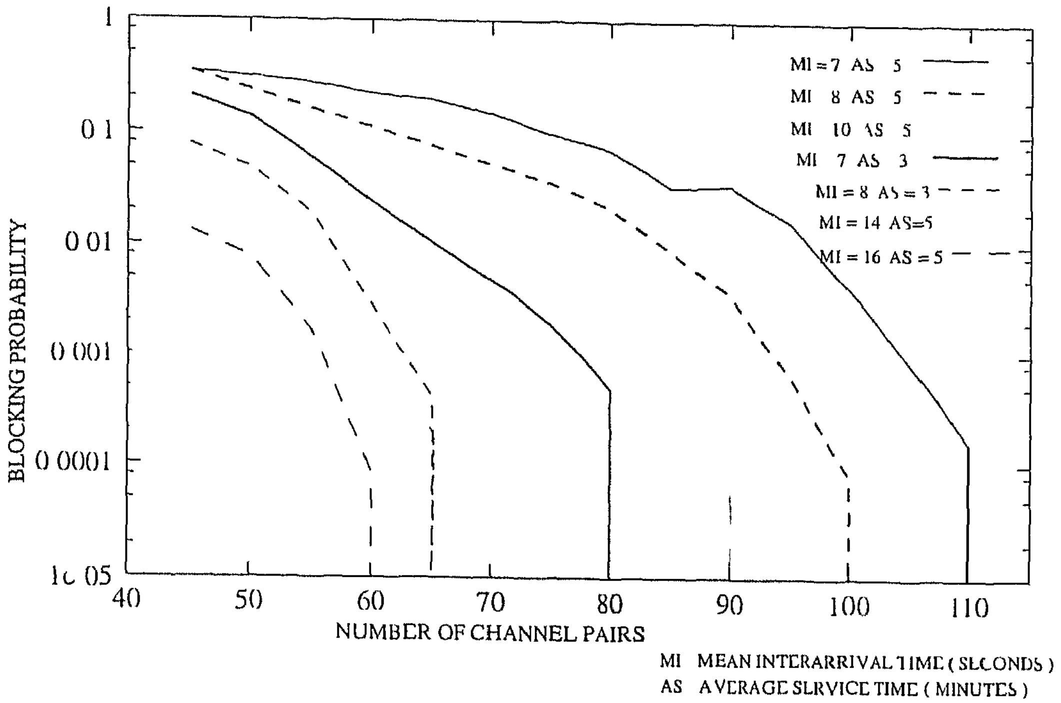


Figure 4 10 The blocking characteristic for different values of mean interarrival time and call holding time (AS) and N

equal to 5 minutes

The characteristics for average traffic carried for the average call holding time of 5 minutes, versus the number of channel pairs (servers) with mean interarrival time as a parameter are illustrated in Fig 4 11 The average traffic carried is observed to remain constant for values of mean interarrival time greater than 14 seconds For a lower value of N, with mean interarrival time of 7 seconds and 8 seconds, the traffic carried increases steadily as N increases These curves then saturate to a constant value, for values of N greater than 80 The characteristics are of this nature because for low values N, a significant number of calls are blocked for these values of mean interarrival time The offered traffic is larger than what the system can carry with out any traffic loss for these values of parameters Moreover as the number of channels increases, the capacity of the system to carry the traffic also increases Hence the fraction of traffic that is lost decreases The channel utilization is also found to be higher for N less than 80 with mean interarrival time of 7 seconds The variation in channel utilization with N for different values of mean interarrival time is illustrated in Fig 4 12

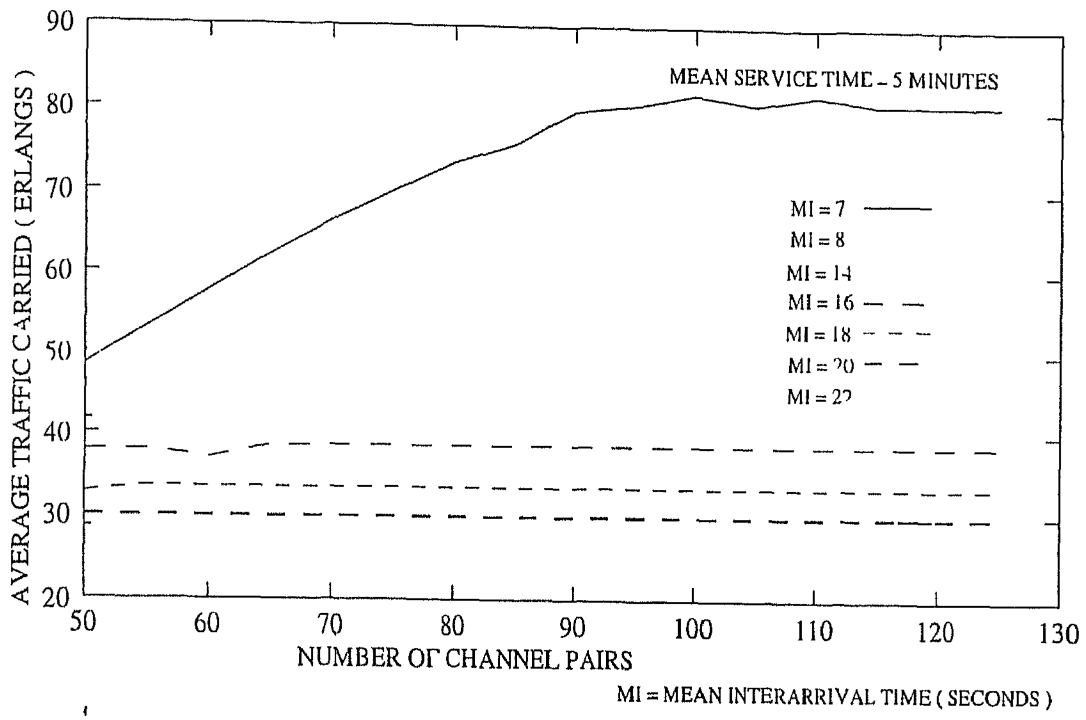


Figure 4 11 Characteristics for average traffic carried for the average call holding time of 5 minutes, versus the number of channel pairs (servers) with mean interarrival time as a parameter

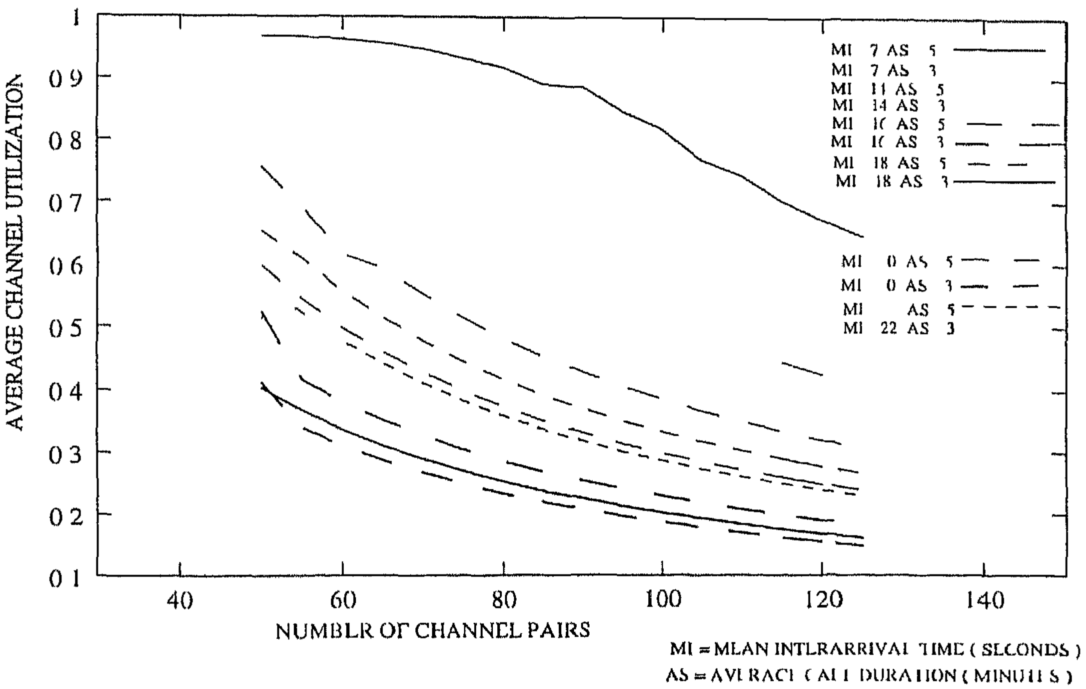


Figure 4 12 The variation in channel utilization with N for different values of mean interarrival time

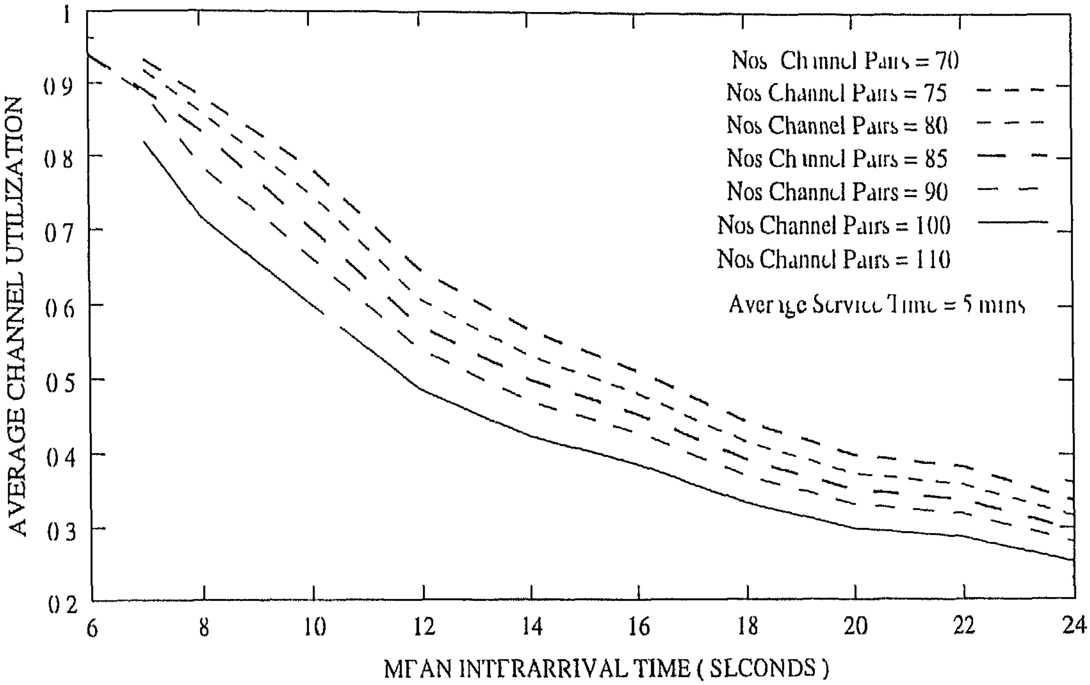


Figure 4 13 Variation of channel utilization with the mean interarrival time for call holding time of 5 minutes

For values of mean interarrival time greater then 7 seconds the channel utilization is found to be low It decreases sharply as N increases from 50 to 90, beyond that the decrease is less sharp The channel utilization is found to be excellent for a mean interarrival time of 7 seconds with 5 minutes average call holding time We also notice a substantial difference in the characteristic of channel utilization for the value of average call holding time of 5 minutes, and that for 3 minutes The variation of channel utilization with the mean interarrival time can be observed in the Fig 4 13 and Fig 4 11, for the average call holding time of 5 minutes and 3 minutes respectively As it should be expected, we clearly observe that better channel utilization characteristics are obtained for the average call holding time of 5 minutes than that for 3 minutes The channel utilization is observed to decrease steadily as the mean interarrival time varies from 6 seconds to 24 seconds This observation can be made for both the figures The channel utilization characteristics is found to saturate towards a maximum value of 1 for N equal to 70 and average call holding time of 5 minutes, for values of mean interarrival time less than 8 seconds These parametric values identify the traffic conditions for which the channel utilization is maximum

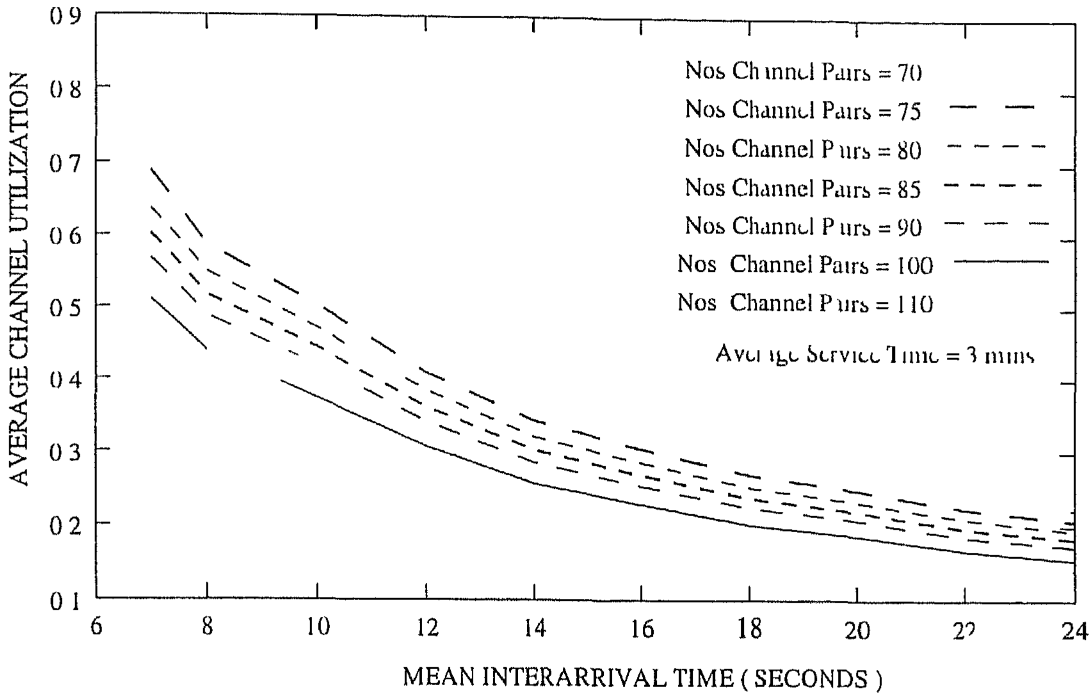


Figure 4 14 Variation of channel utilization with the mean inter arrival time for call holding time of 3 minutes

Fig 4 15 illustrates the variation of average traffic carried with a increase in the mean inter arrival time, for average call holding time of 5 minutes and 3 minutes As it should be expected the traffic carried is higher for the average call holding time of 5 minutes It decreases steadily with the increase in mean interarrival time The fall is found to be sharper for lower values of mean interarrival time Also the fall in it's value is greater for average call holding time of 5 minutes, than that for 3 minutes

The average call set up time value is found to be rather high on account of multiple satellite hops[3] encountered by the signaling units during the call set up phase The characteristic for the average call set up time is shown in Fig 4 16 The value of average call set up time is largely contributed due to the path delay of 6 hops for HUB originated and 7 hops for mobile originated calls Transmission queuing, and processing delays[4] also contribute to this It is observed to decrease sharply as the mean inter arrival time increases from 6 seconds to 10 seconds for it's higher values the call set up time is found to remain relatively constant This is observed because for the low values of mean interarrival time the queuing delay for the signaling units

[3]Each hop contributes to a path delay delay of 480 msecs

[4]Processing delays are neglected in the simulator

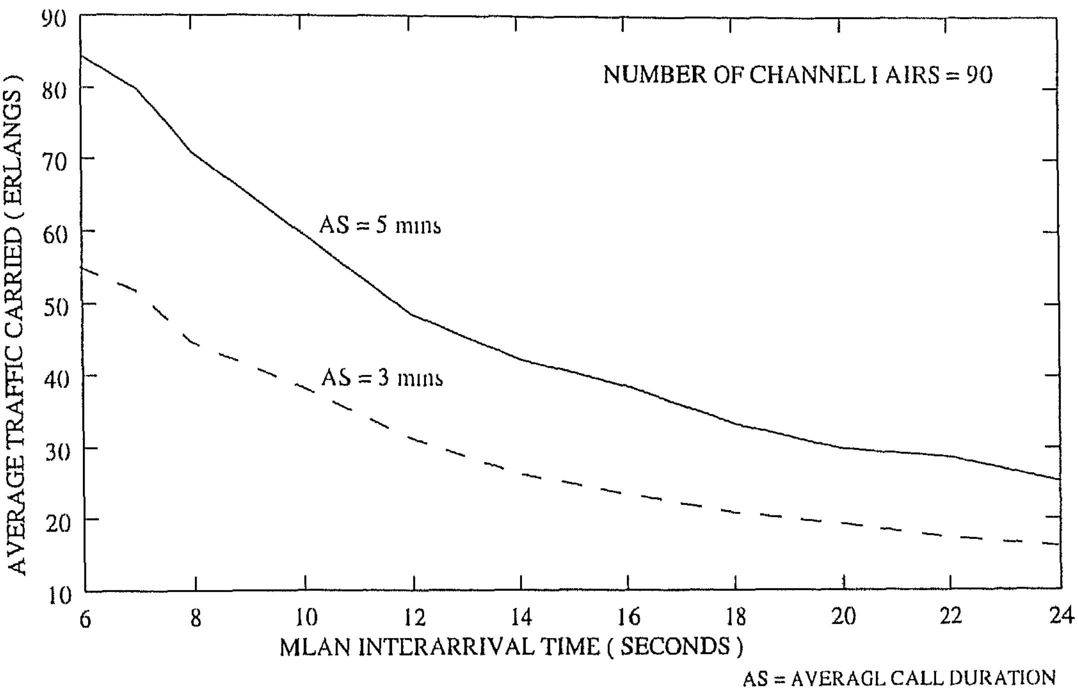


Figure 4.15 Variation of average traffic carried with a increase in the mean interarrival time, for average call holding time of 5 minutes and 3 minutes

in the TDM channel is larger. Also, the delay in the Slotted ALOHA channel is greater for these values of mean interarrival time. Thus in addition to the path delay, the queuing delay forms a greater fraction of the total call set up time. The queuing delay however becomes insignificant for higher values of mean interarrival time. For these values of mean interarrival time, the path delay, transmission delay and the processing delay contribute for the total call set up time. Since path delay and transmission delays are constant parameters, the value of average call set up time is found to be nearly constant for higher values of mean interarrival time. Also, it's variation is comparatively low as compared to it's absolute value.

Fig. 4.17 shows the variation of Slotted ALOHA throughput for the range of mean interarrival time of 6 seconds to 24 seconds. The throughput tends towards a maximum value of 0.1, for mean interarrival time of 6 seconds. The throughput is observed to increase steadily as the mean interarrival time decreases from 24 seconds to 8 seconds. It ceases to rise steadily for mean interarrival time less than 8 seconds and assumes a maximum value of 0.1, for the mean interarrival time of 6 seconds. For values lower than 6 seconds, the Slotted ALOHA is found to be unstable in

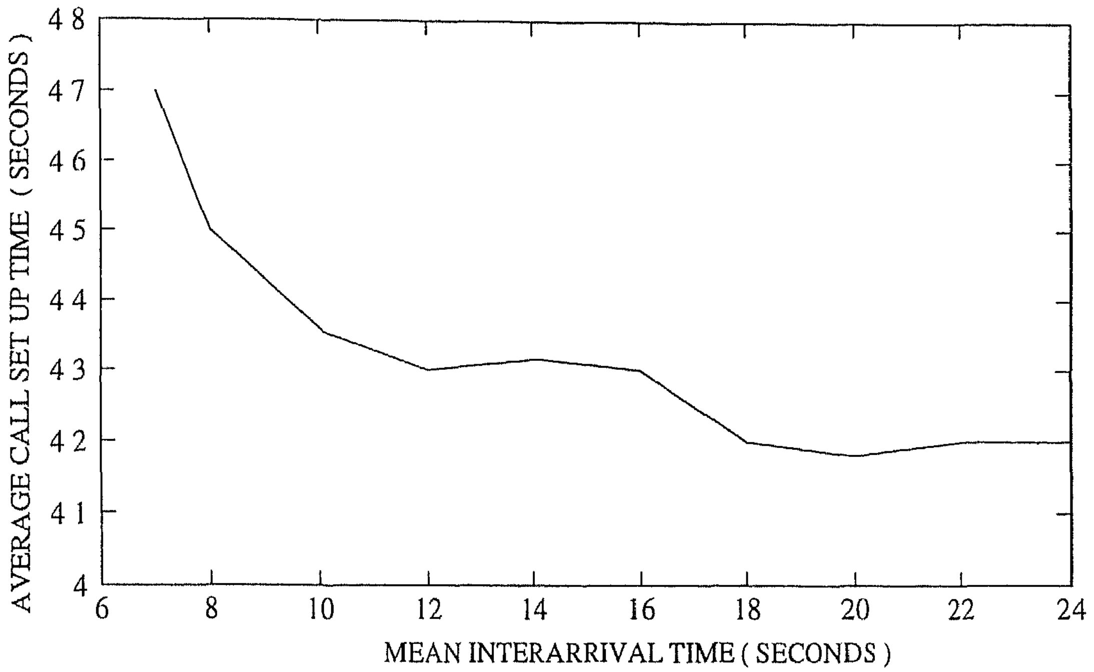


Figure 4 16 Characteristic for the average call set up time

operation The absolute value of throughput and it's characteristics depends upon the backoff mechanism to be employed In this simulation model we have used a simple uniform back off mechanism The mobile terminals back off after collision and tries to retransmit in one of the next 10 slots chosen randomly by a uniformly distributed random number, which takes values between 1 to 10 In absence of any access control scheme for the mobiles to transmit in the Slotted ALOHA channel, the maximum value of throughput is found to be low[5] This hampers the operation of system for mean interarrival time less then 6 seconds Due to the unstability of Slotted ALOHA the characteristics of the system could not be portrayed for values of mean interarrival time less than 6 seconds

4 4 Timers

The timers used in the system are simulated with their approximate values These timers have been simulated as time to live condition after the transmission of the

[5]This can be compared with that for the Class B type of system where an effective flow control scheme is used

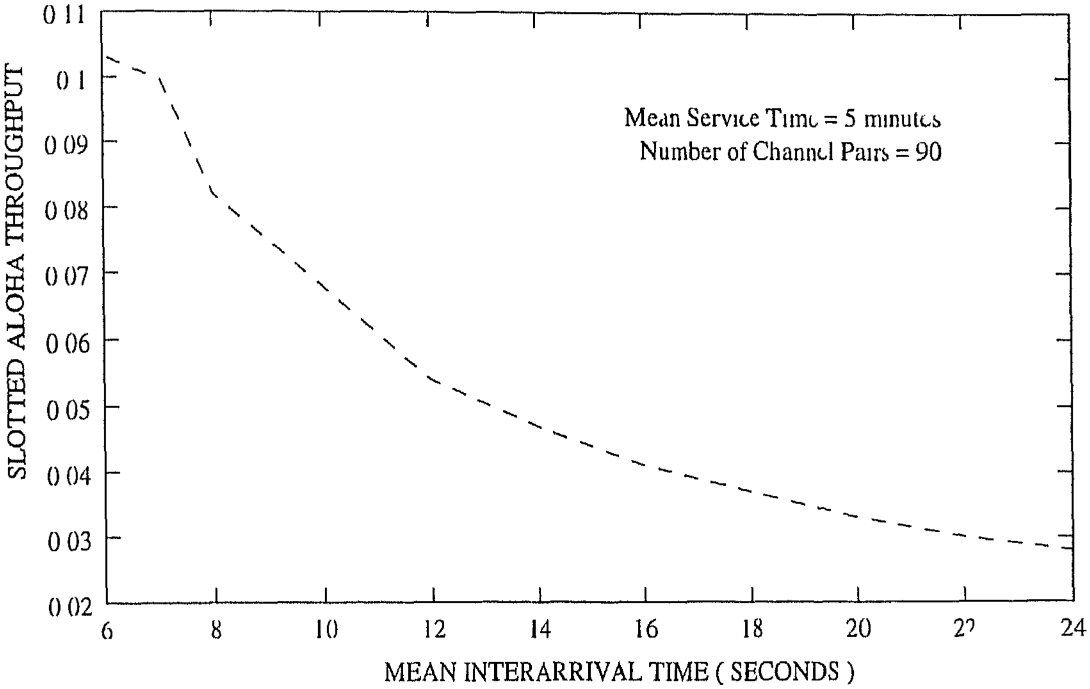


Figure 4 17 Slotted ALOHA throughput characteristics

signaling units, the reply to this is awaited for resetting the corresponding timer The values of the timers have been chosen so as to obtain a reasonably low number of call rejections (typically to be less than 1% of the calls initiated) during the steady state operation of the system The value of timers as used in the simulation model are given below

TH1 **(86H)** This is the value of timer set at the HUB after transmission of "Channel Assignment" signal The value used in the simulator is 3 seconds

TH2 **(95H)** This is the value of timer set at the HUB after transmission of 'Scrambling Vector ACK' signaling unit The value used in the simulator is 1 7 seconds

TM1 **(83H)** This is the value of timer set at the mobile after transmission of 'Access Request' signal The value used in the simulator is 1 7 seconds

TM2 **(8DH)** This is the value of timer set at the mobile after transmission of "Scrambling Vector' signaling unit The value used in the simulator is 1 7 seconds

TM3 (8BH) This is the value of timer set at the mobile after transmission of 'Return Carrier ID' signaling unit. The value used in the simulator is 1.7 seconds.

TH20 (81H) This is the value of timer set at the HUB after transmission of 'Call Announcement' signal. The value used in the simulator is 3 seconds.

TM20 (82H) This is the value of timer set at the mobile after transmission of 'Mobile Response' signal. The value used in the simulator is 3 seconds.

TH21 (86H) This is the value of timer set at the HUB after transmission of 'Channel Assignment' signal for HUB initiated call. The value used for this timer is same as TH1.

For the same signaling units in case of mobile originated calls and HUB originated calls, the timer values used are identical.

Chapter 5

CLASS-B SERVICES : PROTOCOLS AND PROCEDURES

5 1 Introduction

The Class B type of service is a store and forward messaging service using shared channels The HUB acts as the store and forward unit for this service The messages, their acknowledgements the requests for transfer and other signalling protocols will be sent from the HUB to the mobile terminals in a Time Division Multiplexed channel (forward TDM channel) Four parallel Slotted ALOHA channels will be used for carrying the messages and other protocol packets from the mobile terminals to the HUB The messages are transferred after the successful transfer of handshaking signals Class B type of mobile terminals can transfer data to the PSTN data links and telex subscribers Messages can also be transferred from the PSTN data links and telex subscribers to the Class B type of mobile terminals Messages, that could not be forwarded by the HUB shall be stored in the mailbox of the respective mobile or PSTN subscriber These stored messages can be down loaded by the mobile terminals or the PSTN subscribers after sending a query to the HUB The major specifications for this service are

- Messages shall be only in 'ASCII' (TEXT) format

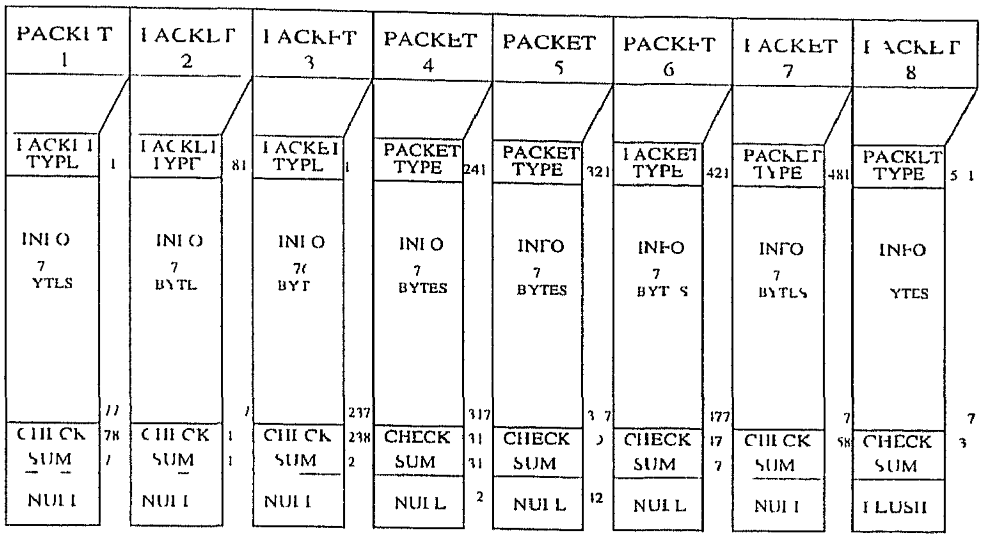


Figure 5.1 TDM channel information field format

These 639 bytes (5112 bits) of data and 1 flush byte (8 bits) are then differentially encoded, scrambled, rate 1/2 FEC coded and interleaved. The frame thus formed shall consist of 10224 bits representing 5112 bits of data and two unique words of 64 bits each. This frame format is shown in Fig. 5.2. The interleaving and coding details are given in [4].

5.2.2 Slotted ALOHA Messaging channels

There will be four parallel Slotted ALOHA channels for carrying the protocol packets from the mobile terminals to the HUB. Each of these channels will operate at 1.2 Kbps and is expected to have a BER of 1×10^{-6}. During each TDM frame time of 8.64 seconds there will be six burst slots. These burst transmissions will be synchronized to the incoming TDM frame slots as shown in Fig. 5.3. Each burst slot will carry a single protocol packet of 80 bytes. The protocol packets will have a structure identical to packets in the forward TDM frame. These 80 bytes (640 bits) of the protocol packet including the last flush byte are differentially encoded, scrambled, rate 1/2 FEC coded and interleaved. The burst thus formed will finally consist of 1280

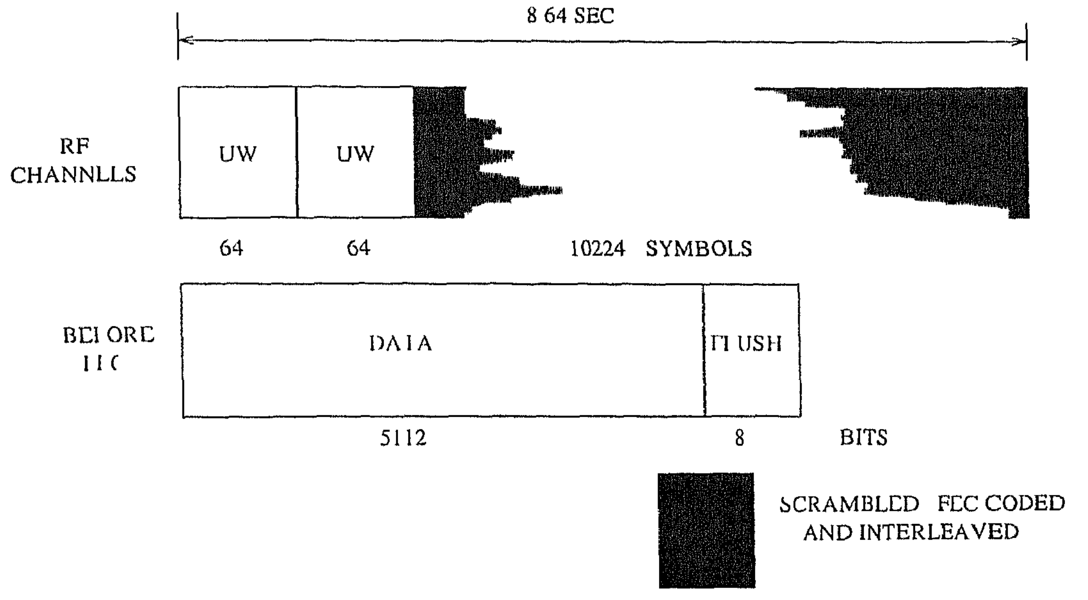


Figure 5 2 TDM channel frame format

bits of information representing 632 bits of data and 8 flush bits, 4 copies of 16 bit unique word and a Bit Timing Recovery pattern of 192 bits for the demodulator at the HUB This burst format is shown in Fig 5 3 The interleaving and coding details are given in [4]

5 3 Protocol Packets Types and Format

The various types of protocol packets used for the Class B type of messaging service are listed in Table 5 1 The specific details for each of these packets and their different fields are given in [4] The generalized format of any protocol packet has been described in Section 5 2

Some of the protocol messages may be of more than one packet, eg the "BB" (01H) and the "Message Received ACK" (07H) messages For these cases multiple packets are transmitted For these protocols, the packets carry the "total number of pages" and the "page number" in their information field The mailbox response may also be in more than one packet, with each packet containing "number of mails" and "mail number" The BB packet will carry the burst slot utilization data for the return Slotted ALOHA channels and the number of users transmitting in these channels This

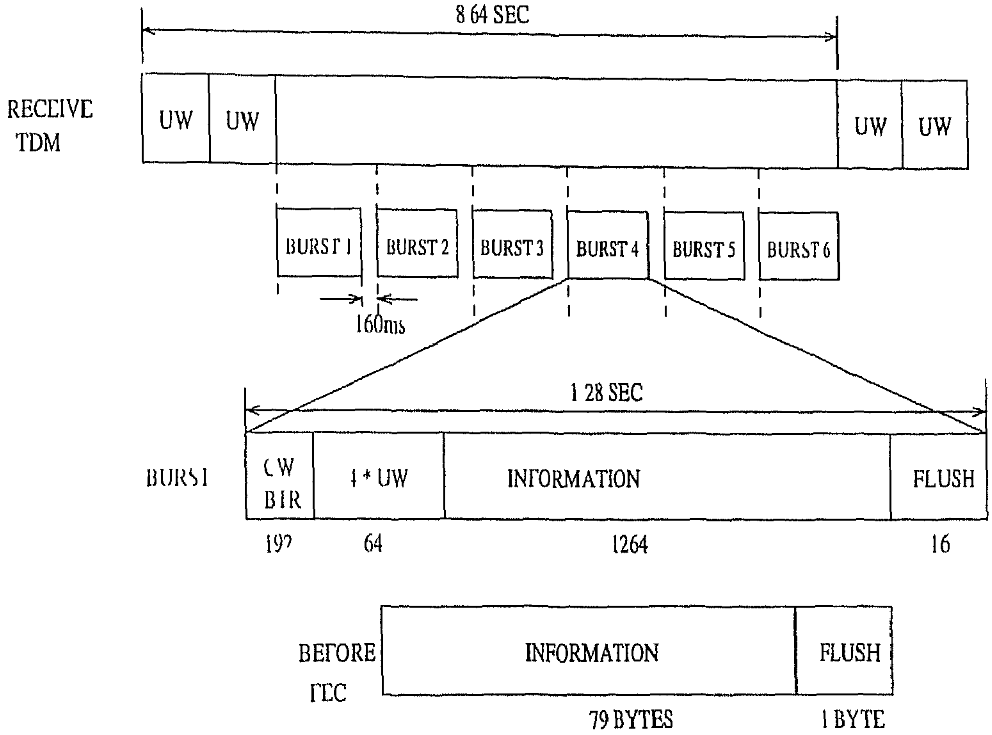


Figure 5 3 Slotted ALOHA channel burst format

data will correspond to the $(N-2)^{nd}$ TDM frame time, where N is the current TDM frame If the corresponding burst slot was utilized, a '1' is set in the bit, otherwise the bit is reset Fill in packets will be transmitted in the TDM channel if no other protocol packet is to be transmitted The TDM channel frames will be filled with protocol packets according to the following order of priority

1 MFR ACK (06H) for priority requests

2 MRA (07H) for priority requests

3 MRA (07H) for normal requests

4 Confirmation (0AH)

5 MRA Request (08H)

6 MFR ACK (06H) for normal requests

7 Message packet (04H)

8 MTR (02H)

PACKET TYPE (HEX)	FUNCTION TYPE	TRANSMIT CHANNEL
0 0	Fill-in Packets	-
0 1	Bulletin Board Packets	TDM
0 2	Message Transfer Request (MTR)	TDM
0 3	MTR Acknowledgement	S-ALOHA
0 4	Message Packets	TDM / S-ALOHA
0 5	Message Forward Request (MFR)	S-ALOHA
0 6	MFR Acknowledgement	TDM
0 7	Message Received ACK (MRA)	TDM / S-ALOHA
0 8	MRA Request	TDM
0 9	Spare	-
0 A	Confirmation Packet	TDM
0 B	Request for Confirmation	S-ALOHA
0 C	Spare	-
0 D	Spare	-
0 E	Spare	-
0 F	Spare	-
1 0	Mailbox Request	S-ALOHA
1 1	Mailbox response	TDM
1 2	Download Request	S-ALOHA
1 3	Download ACK	TDM
1 4	Spare	-
1 5	Spare	-
1 6	Spare	-

Table 5.1 : Packet Types For CLASS-B Service

9. Download ACK (13H)
10. Mailbox response (11H)
11. BB (01H)
12. Fill-in (00H)

When a transaction is initiated with any mobile terminal it is marked as 'BUSY' at the HUB. Messages are transferred to or from the mobile terminals only if they are not marked as 'BUSY' in the "BUSY list". In the system, flow control is present only at the message level. The messages to be transferred are broken into packets of 64 bytes each and transmitted as the information field in the protocol packet "Message packet" (04H). There will be only one message transfer in a single transaction.

5.4 HUB to Mobile Message Transfer : Protocols and Procedure

The sequence of packets exchanged for transferring messages from the HUB to the mobile terminal is depicted in Fig. 5.4. The transfer begins by the HUB checking for the 'BUSY' status of the mobile. If it is not already in the "BUSY list", then the HUB places the mobile terminal in this list. It then sends a "Message Transfer Request" MTR (02H) and waits for the MTR ACK (03H) from the mobile terminal. If the acknowledgement is not received within a certain time out, the MTR packet is retransmitted with 'retry' field set. At most three such retries are made. If no response is received after the third retry, the message is stored in the mailbox and the mobile terminal is removed from the busy list.

On receiving a MTR from the HUB, the mobile terminal selects a return Slotted ALOHA channel and sends back the MTR ACK (03H) message. The priority here is given to transfers from the HUB to the mobile. In case the mobile terminal has initiated a transfer to the HUB, this transfer is put on hold and the mobile proceeds with the receive transaction. The mobile terminal then waits for receiving the message packets and the "Message received ACK Request" MRA Request (08H), for the

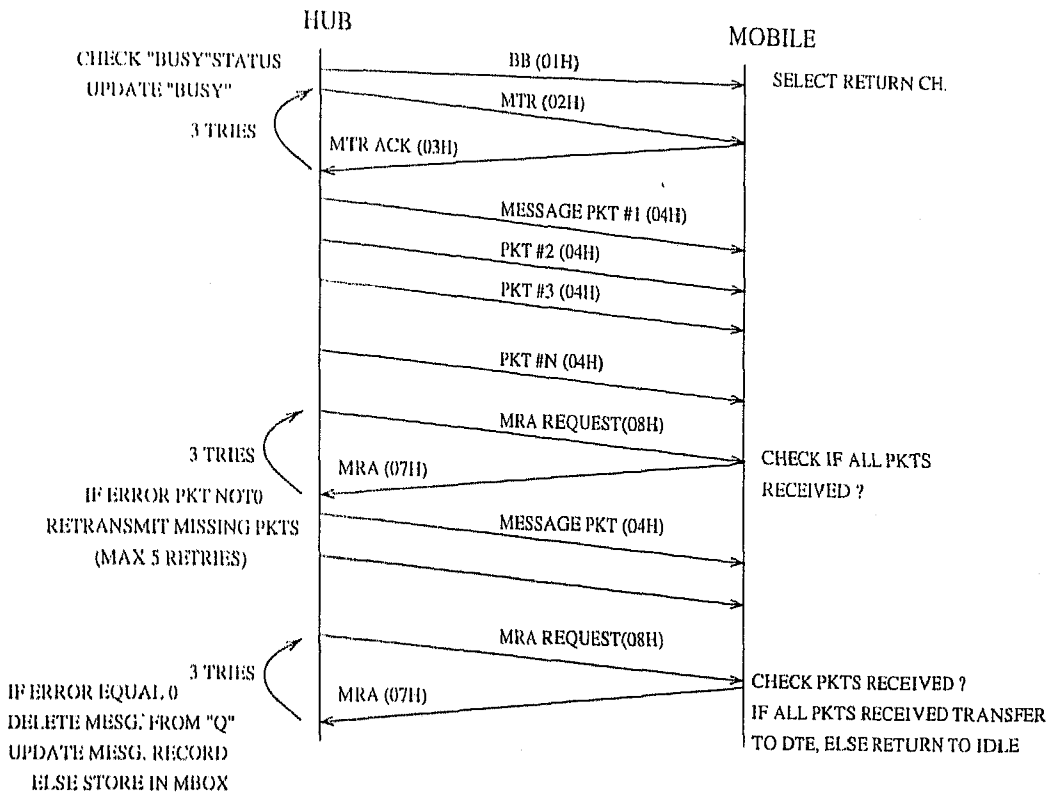


Figure 5.4: HUB to mobile message transfer sequence

next (N+10) TDM frame time, (Here N is the expected number of message packets to be received by the mobile terminal). On failure to receive the MRA Request within the above specified time, the mobile will abandon the receive transaction.

On receiving a MTR ACK (03H) from the mobile terminal, the HUB initiates the transfer of message packets. The message packets (04H) are transmitted in the TDM channel according to the above mentioned priority rules. There after the HUB sends a MRA Request (08H) to obtain the "Message Received Acknowledgement" MRA (07H) from the mobile terminal. If the MRA is not received within a specified timeout the MRA Request is retransmitted. A maximum of three such retransmissions will be made. The message transfer will be aborted if the HUB fails to get the MRA within the three retransmissions; in this case the message will be deposited in the mailbox and the mobile terminal will be removed from the "BUSY list".

The mobile terminal responds with MRA (07H) after receiving a MRA Request (08H). The MRA will contain the information regarding number of packets in error and the packet numbers that were received in error. Specific packet numbers

of the missing/error packets are sent in the MRA. If the number of missing packets is more than 33, the MRA will be in more than one page. If the number of missing/error packets exceeds 50, then "99" will be sent in the "error packet" field and the transfer will be aborted. If there are no packets missing or in error, then "00" is sent in the "error packet" field. This will indicate a successful transfer of the message. In this case after waiting for a small time 'T' the mobile returns to the "idle" mode.

On receiving the MRA (07H) from the mobile terminal, the HUB retransmits the packets in error, (if there are any and if their number is less than 50). If there are no packets to be retransmitted , the message transfer is completed successfully and the mobile is removed from the "BUSY list". The maximum number of retransmissions allowed for packets in error is five. If the number of packets in error is greater than 5.or if the message is not successfully transferred even after maximum retries, then it will be stored in the mailbox, the message transfer will be aborted and the mobile removed from the "BUSY list".

5.5 Mobile to HUB Message Transfer : Protocols and Procedure

A message transfer from the mobile to the HUB is processed only if no request is initiated for the transfer of message from the HUB to the same mobile terminal. The HUB to mobile message transfer sequence is shown in Fig 5.5. The mobile terminal waits for a BB packet to check for the Slotted ALOHA channel activity data and the number of users in each Slotted ALOHA channel. The mobile terminal selects a Slotted ALOHA channel for which there are less than three users and not all bits corresponding to burst slot activity are set to '1'. If no such return channel is available, the mobile terminal waits for the next BB packet. The mobile continues to wait until the above specified conditions are satisfied. After the Slotted ALOHA channel is selected, the mobile terminal transmits the "Message Forward Request" MFR (05H) packet and waits for the MFR ACK (06H) packet until a certain time period. If the MFR ACK is not received within this timeout period, the MFR (05H) is retransmitted. A maximum of three such retries is made for MFR. If the MFR

ACK is not received within three retries, the message transfer is aborted.

The HUB on receiving the MFR packet, checks for the presence of the mobile terminal in the "BUSY list" . It validates the request and checks if there are less than three users in the selected Slotted ALOHA channel. If all these conditions are satisfied, the mobile terminal is placed in the "BUSY list". The HUB then transmits the MFR ACK three times with the 'confirmation status' as "y". If any of the above mentioned conditions are not satisfied the MFR ACK is transmitted with the 'confirmation status' and 'cause indication' set to indicate the appropriate reason. After the transmission of MFR ACK with 'confirmation status' set to "y", the HUB is expected to wait for for (N+2) TDM frame times for receiving all the message packets, where N is the number of message packets expected by the HUB. If the message packets are not received within this timeout, the message transfer is aborted and the mobile terminal is removed from the "BUSY list". The mobile on receiving the MFR ACK with 'confirmation status' as "y" starts the transmission of message packets in sequence. One of the six burst slots is selected randomly to transmit each message packet. If the 'confirmation status' is not received as "y", the message transfer is aborted.

The HUB transmits three copies of the MRA (07H) packet after receiving the expected number of message packets. The MRA packet will indicate the number of packets in error and the message packet numbers that are in error. The same set of rules as described in Section 5.4., for the HUB to mobile transfer will be used for encoding the details of these packets in error in the MRA packet. The HUB then waits for a certain time out for retransmission of missing packets if any. The maximum number of retransmissions is limited to five. If no packet is in error or no packet is lost, then the message transfer is successful and the mobile terminal is removed from the "BUSY list". HUB then tries to forward the message to the PSTN subscriber.

The mobile terminal waits for a certain time out period for receiving the MRA (07H) from the HUB, after transmitting all the message packets. If no MRA is received within this timeout, the transfer is aborted and the status is reported to the mobile DTE. The mobile terminal will retransmit the packets in error, if there are any, and

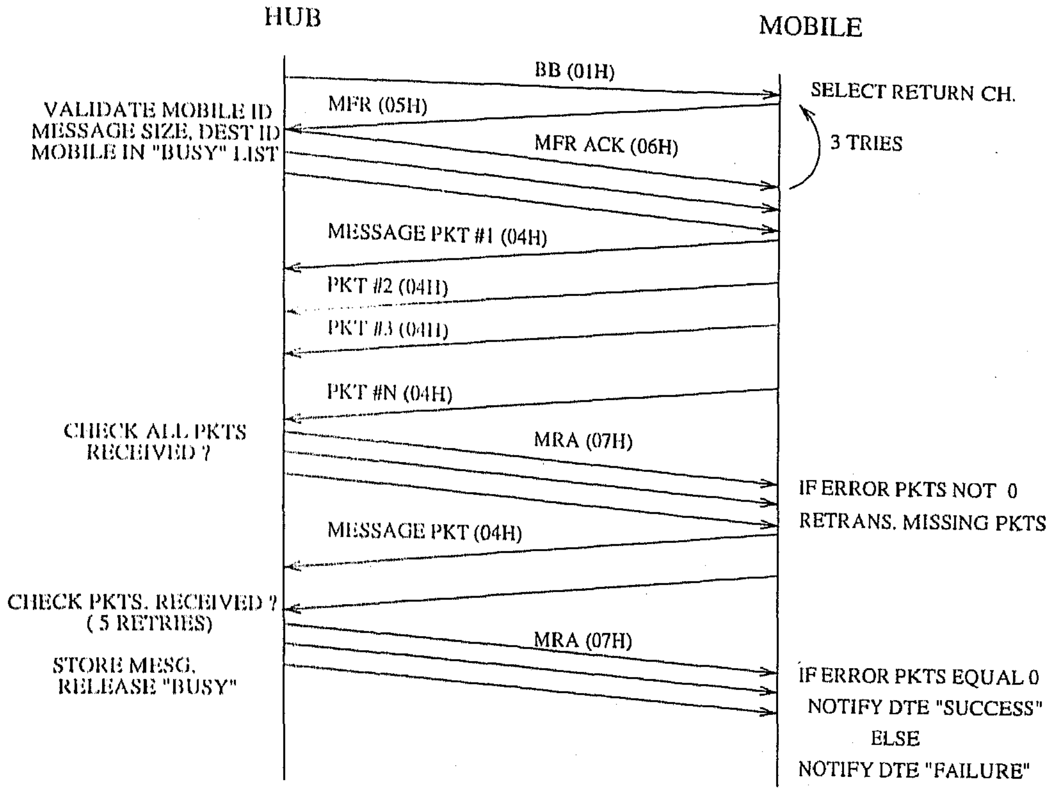


Figure 5.5: Mobile to HUB message transfer sequence

if their number is less than 50. If there are no packets in error, the message transfer is successfully completed. If the number of packets in error are greater than 5, or if the message is not successfully transferred even after maximum number of retries, the transfer is aborted and the status reported to the mobile DTE.

5.6 Mailbox Service

The Class-B type of service is a store and forward service. The messages for the mobile terminals and the PSTN subscribers are stored at the HUB. The HUB tries to forward these messages to the respective destinations as described in the previous sections. The messages that could not be forwarded are stored in the respective mail boxes. The mobile terminals can send a query to their mailbox and down load the messages collected in the mailbox. The protocols and procedure used for this down loading of mails are described in this section.

A maximum of 9 messages shall be stored in the mailbox of the respective mobile terminals. The mailbox down loading procedure for the mobile terminals is shown in the Fig. 5.6. The mobile terminal first selects a Slotted ALOHA channel as per the access rules described in the previous section. The "Mailbox Request" (10H) is sent in the Slotted ALOHA channel selected, to check if there are mails present in the mailbox. The mobile terminal will retransmit the "Mailbox Request" if there is no response within a certain timeout. The maximum number of retries for this are limited to three.

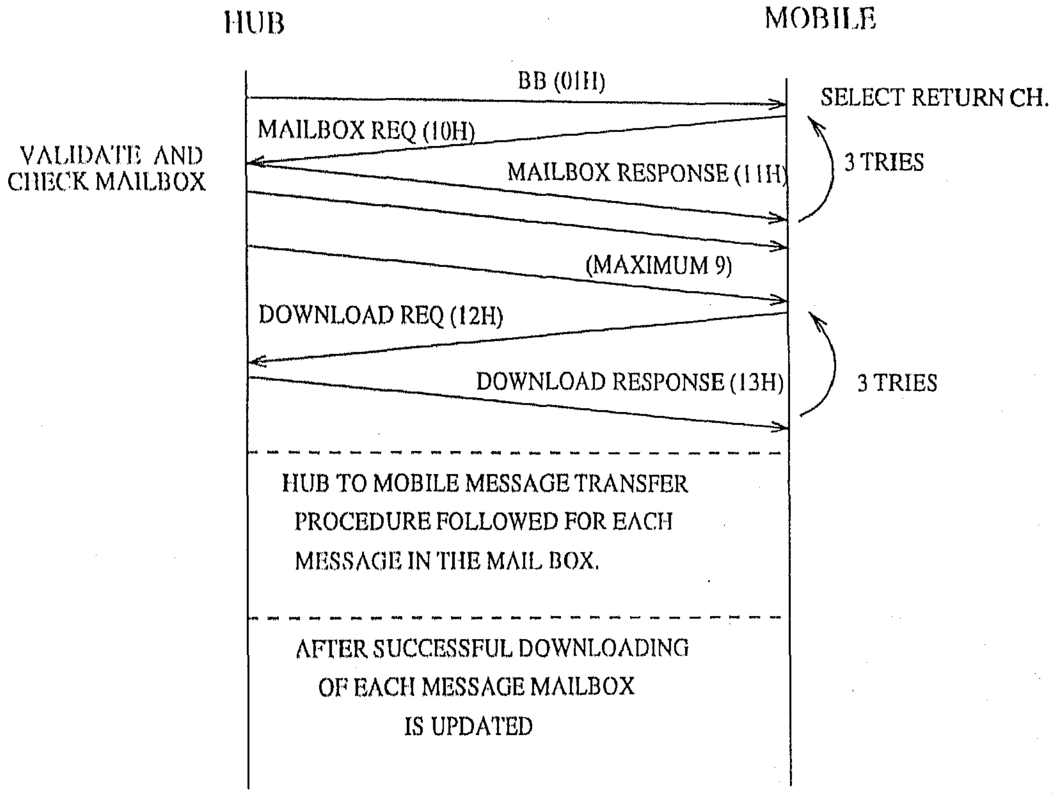


Figure 5.6: Mailbox access sequence for mobile users

On receiving the "Mailbox Request" the HUB responds with the "Mailbox Response" (11H) packet. The HUB sends three copies of this response. The number of packets in the response will be the same as the number of messages present in the mailbox.

On receiving the "Mailbox Response" the mobile terminal tries to down load the messages by sending the "Download Request" (12H) in the Slotted ALOHA channel. The mobile terminal then waits for some time to receive the "Download ACK" (13H)

from the HUB. The "Download Request" is retransmitted if the "Download ACK" is not received within the timeout period. The maximum number of such retransmissions will be three. On receiving the "Download ACK", the mobile terminal waits for the MTR packet from the HUB, signifying the transmission of messages from the HUB to the mobile.

On receiving a "Download request" the HUB sends back a "Download ACK". The HUB then starts down loading the messages to the mobile. After the message has been successfully transferred, the mailbox is appropriately updated. The same process will be repeated for downloading each message in the mailbox.

Chapter 6

CLASS-B: SIMULATION MODEL AND ANALYSIS

6.1 Simulation Model

The class B type of system described in the previous chapter can be modeled as shown in Fig. 6.1. Using this model a discrete time event type of simulator of the above mentioned system has been developed. The simulator has been used to analyze the performance of system, verify the various protocols and find typical values for it's operating parameters. We developed the discrete time event type of simulator using 'C' as the programming language. The functional blocks of the model have been emulated in the simulator by employing various data structures. The various event routines control the operation of the system as it evolves over time and implements the functions, that the system performs during it's operation. The major components of the system and the data structures used to implement them are summarized below.

6.1.1 Functional Blocks

A basic data structure has been defined that contains all the major fields to identify and process the message transfers in either direction. This data structure is shown in Fig. 6.2.; we call this cust. The pointers to objects of this class are passed through queues, representing the channels for communication between the HUB and

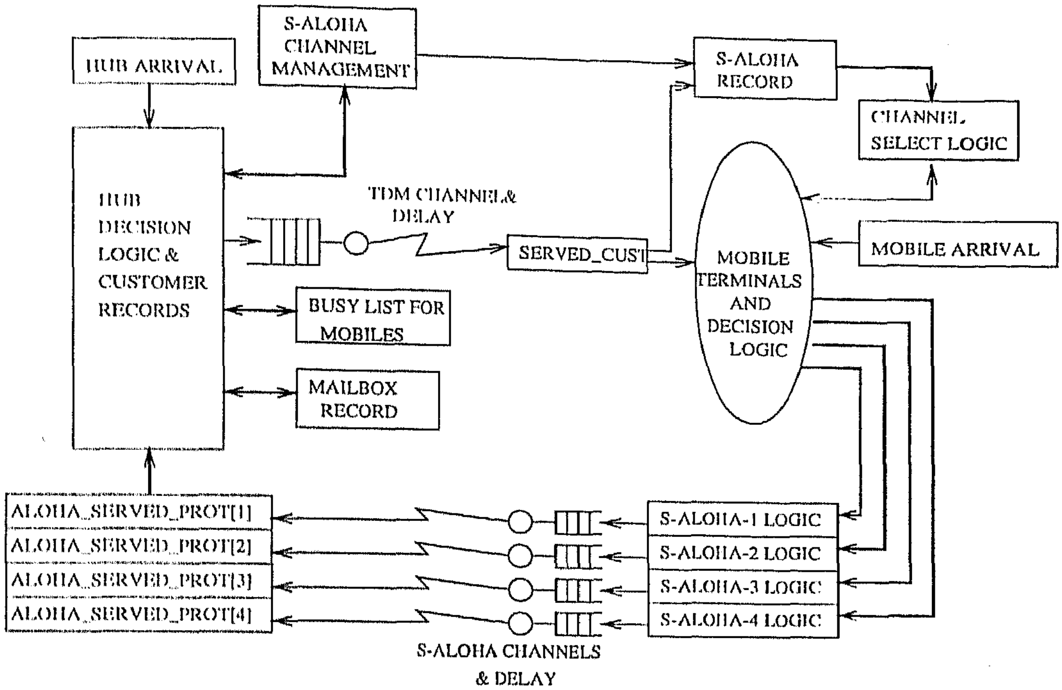


Figure 6.1: Model for Class-B type of system

the mobile terminals.

HUB The data structures representing the components of HUB are

customer This is the representative list of customers at the HUB, elements of array customer shall contain the pointer to particular object of type cust. This object has various details of the message transfer and the associated protocols.

mob_cust_busy This is the list of status of mobile terminals. These can be BUSY or FREE depending on whether the corresponding mobile terminal is busy or free.

mailbox This array contains the list of number of mails for each mobile customer.

mail_mesg Maintains the list of each message size for all messages of all mobile customers, stored in the mailbox.

burst_slot_aloha Contains burst slot utilization data for the Slotted ALOHA channels for $(N-2)^{nd}$ TDM frame if current TDM frame number is N.

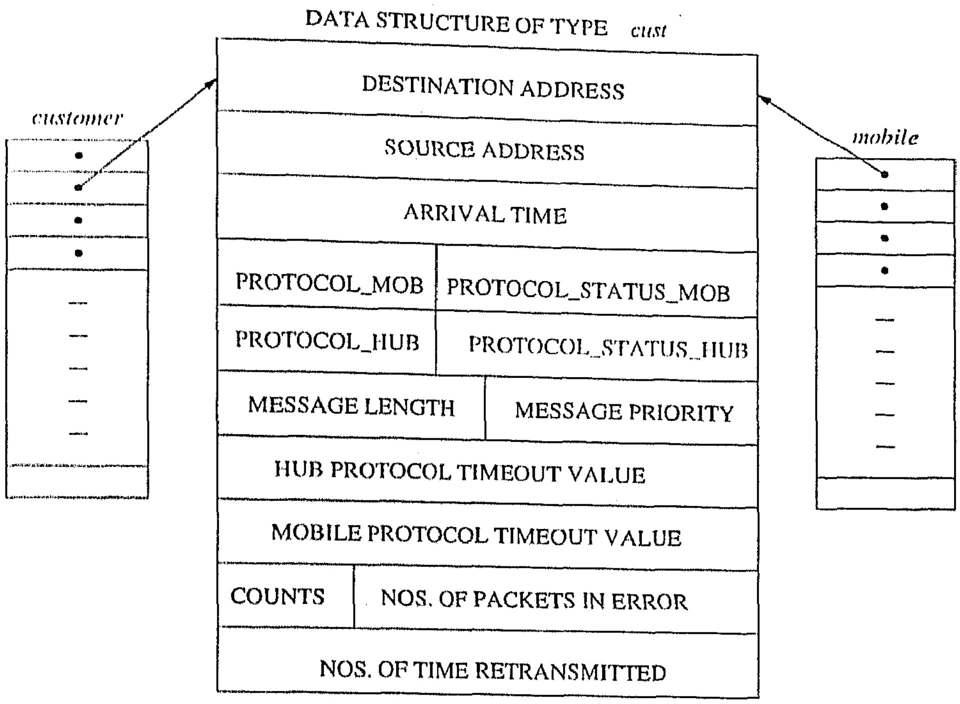


Figure 6.2: The data structure cust

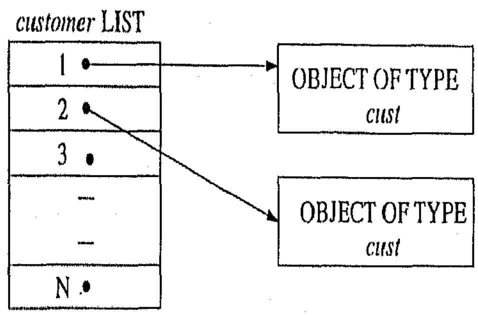


Figure 6.3: Data structures for representing the customers at HUB

update_burst_slot_aloha Contains burst slot utilization data for the Slotted ALOHA channels for the current TDM frame.

update1_burst_slot_aloha Contains burst slot utilization data for the Slotted ALOHA channels for $(N-1)^{st}$ TDM frame if current TDM frame number is N.

nos_mob_aloha Contains the number of mobile customers currently transmitting in a Slotted ALOHA channel. This parameter is used by the mobile customers for selecting a Slotted ALOHA channel before they start to transmit. This is updated and made available to mobile customers only when there are no packets to be transmitted in TDM queue at HUB i.e.

when a Bulletin Board packet is transmitted.

nos_mob_aloha_local This contains the number of mobile customers currently transmitting in Slotted ALOHA channels. This is maintained at the HUB and is constantly updated.

Time Division Multiplexed (TDM) Channel The TDM channel is simulated using following data structures.

que1 Maintains list of packets to be transmitted in TDM channel, it implements the queue for TDM channel. Each element can contain pointers to objects of type cust.

pkt_priority_list Maintains the list of various protocol packets in the decreasing order of priority, that can be transmitted in TDM channel.

server1_status Maintains the status of server for the TDM queue at the HUB.

served_cust This contains pointers to protocol unit serviced from the TDM channel queue. It is used to simulate the channel delay. It is also used to implement the decision logic for the packets that arrive at the mobile unit.

Mobile Terminals Following is the the data structure representing the mobile terminals .

mobile It is the representative list of mobile customers. It implements each mobile unit along with mob_list, which contains the status of mobile unit and the Slotted ALOHA channel through which it shall transmit.

Slotted ALOHA channels The four Slotted ALOHA channels are simulated using two arrays.

pointers Contains list of pointers to protocol units to be served by Slotted ALOHA channel.

transmit_time This contains the list of corresponding time, when the mobile units are expected to attempt transmission of respective protocol units.

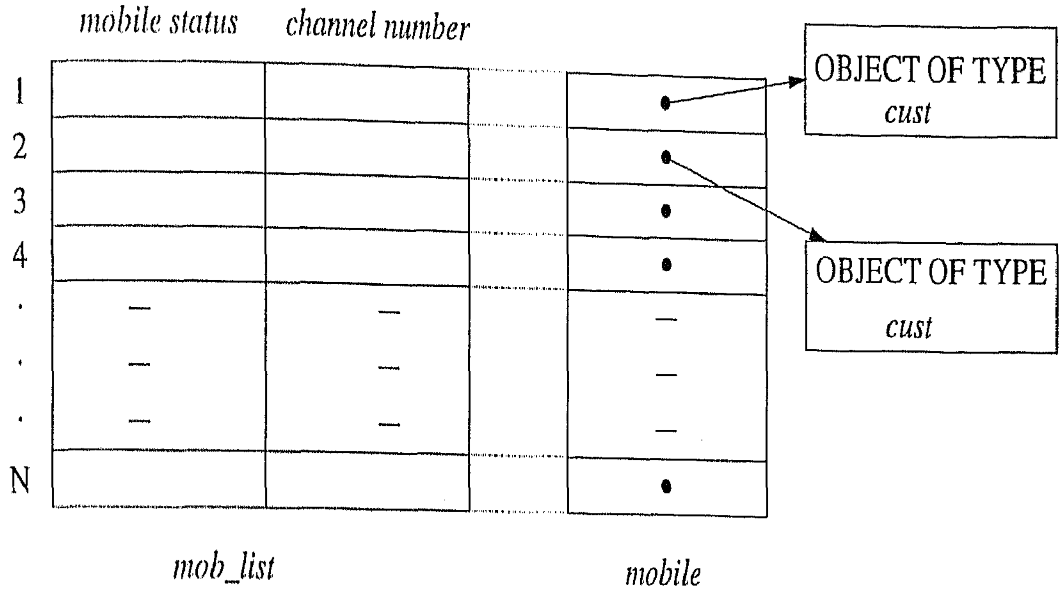


Figure 6.4: Data structures for representing the mobile customers

The Slotted ALOHA is implemented by looking up the transmit_time table at every slot beginning, and if there are more than one entry having values less than current value of time, corresponding action is taken for the collision condition. If only one entry is present then it is served by entering the pointer value in corresponding aloha_served_prot. Then the pointers and transmit_time tables are updated.

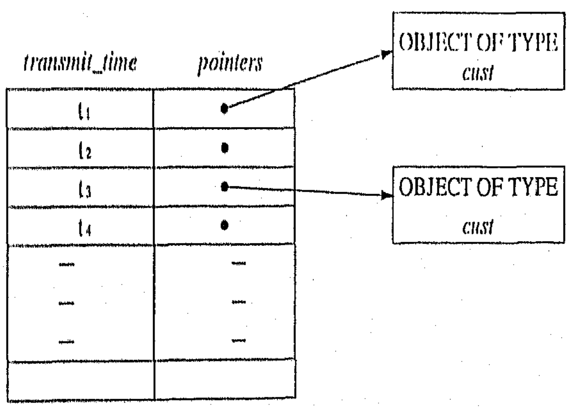


Figure 6.5: Arrays for simulating the Slotted ALOHA channel

6.1.2 System Parameters

MEAN_INTER   The mean interarrival time of customers arriving at the HUB.

MOB_MEAN_INTER The mean interarrival time of mobile terminals, requesting message transfer.

MOB_MEAN_INTER_DOWNLOAD Mean interarrival time of mobile terminals, requesting for "Mailbox Request" and subsequent "Down load Request", for each mail present in mobile mail box at the HUB.

The various interarrival times [5] have exponential distribution with the means represented by the parameters described above. The customers requesting for message transfer generates the message packets. The number of packets to be transferred are uniformly distributed between 1 to 500 since maximum size of a message can be 32 Kbytes and each packet can carry 64 bytes of data [4].

The system parameters include the various timers that control the time out of protocols during message transfers. These timers are represented by

- TIMEOUT_VAL_TWO (02H)
- TIMEOUT_VAL_EIGHT (08H)
- TIMEOUT_VAL_SIX (06H)
- TIMEOUT_VAL_SEVEN (07H)
- TIMEOUT_VAL_FOUR (04H)
- TIMEOUT_VAL_FIVE (05H)
- TIMEOUT_VAL_TEN (10H)

The characters in brackets indicate the protocols after transmission of which these timers are set. The current value of time added to these timer values gives the time to live values for the corresponding protocol packets. This time to live value is placed in the hub_timeout or mob_timeout field of respective protocol packets depending on weather HUB is transmitting or the mobile unit is transmitting the corresponding protocol packet.

6.1.3 Performance Parameters

Parameters that represent the performance of the system in steady state, have been measured by conducting the simulation run for different sets of system parameters. The performance parameters that are likely to represent the steady state characteristics of the system are summarized below.

Mean data transfer rate for HUB to mobile message transfer This is determined by finding the ratio of the total number of data bits transferred from HUB to mobile to the total time period required required for the transfer.

Mean data transfer rate for mobile to HUB message transfer This is determined by finding the ratio of the total number of data bits transferred from mobile to HUB to the total time period required for the transfer.

Average delay per data packet for HUB to mobile message transfer This is determined by taking the ratio, of the sum, of time delays for all the message transfers from HUB to mobile, to the sum of the total number of data packets transferred. The message transfer delay is measured as the time that elapses since a request for the message transfer i.e. MTR is transmitted, till the acknowledgement for successful transfer of all the message packets has been received.

Average delay per data packet for mobile to HUB message transfer This is determined by taking the ratio, of the sum, of time delays for all the message transfers from mobile to HUB, to the sum of the total number of data packets transferred. The message transfer delay is measured as the time that elapses since a request for the message transfer i.e. MFR is transmitted till the acknowledgement for successful transfer of all the message packets has been received.

Slotted ALOHA throughput Throughput has been measured for the four return Slotted ALOHA channels. These Slotted ALOHA channels have been labeled as aloha-1, aloha-2, aloha-3, aloha-4.

Besides these major parameters some other parameters that have been measured and tabulated in the data files are.

- Mean message delay for HUB to mobile message transfer, for messages of size 1 to 99 packets, 100 to 199 packets, 200 to 299 packets, 300 to 399 packets, 400 to 499 packets, has been measured separately, for each of the message size groups.
- Mean message delay for mobile to HUB message transfer, for above mentioned message size groups, have been measured separately.
- Maximum delay for HUB to mobile message transfer.
- Maximum delay for mobile to HUB message transfer.
- Average delay for HUB to mobile message transfer.
- Average delay for mobile to HUB message transfer.
- Number of message transfers aborted due to time out of various timers.

6.2 Flow Charts

A broad outline of the flow of logic followed for the simulation is shown in the flow charts given in Fig. 6.6.

6.3 Analysis

The TDM channel at the HUB and the Slotted ALOHA channels are modeled as single server queues with arrivals from multiple inputs, as can be seen in Fig. 6.1. The approximate analysis for these queues is given in Appendix B. The calculations have been carried out to find the values of mean interarrival time (at the HUB and at the mobiles), such that the necessary condition for stable operation of queues is satisfied. For this condition, assuming no flow control on messages, the mean interarrival time at the HUB and for mobiles is found to be 4.6 minutes and 7.74 minutes respectively, for a mean message size of 250 packets.

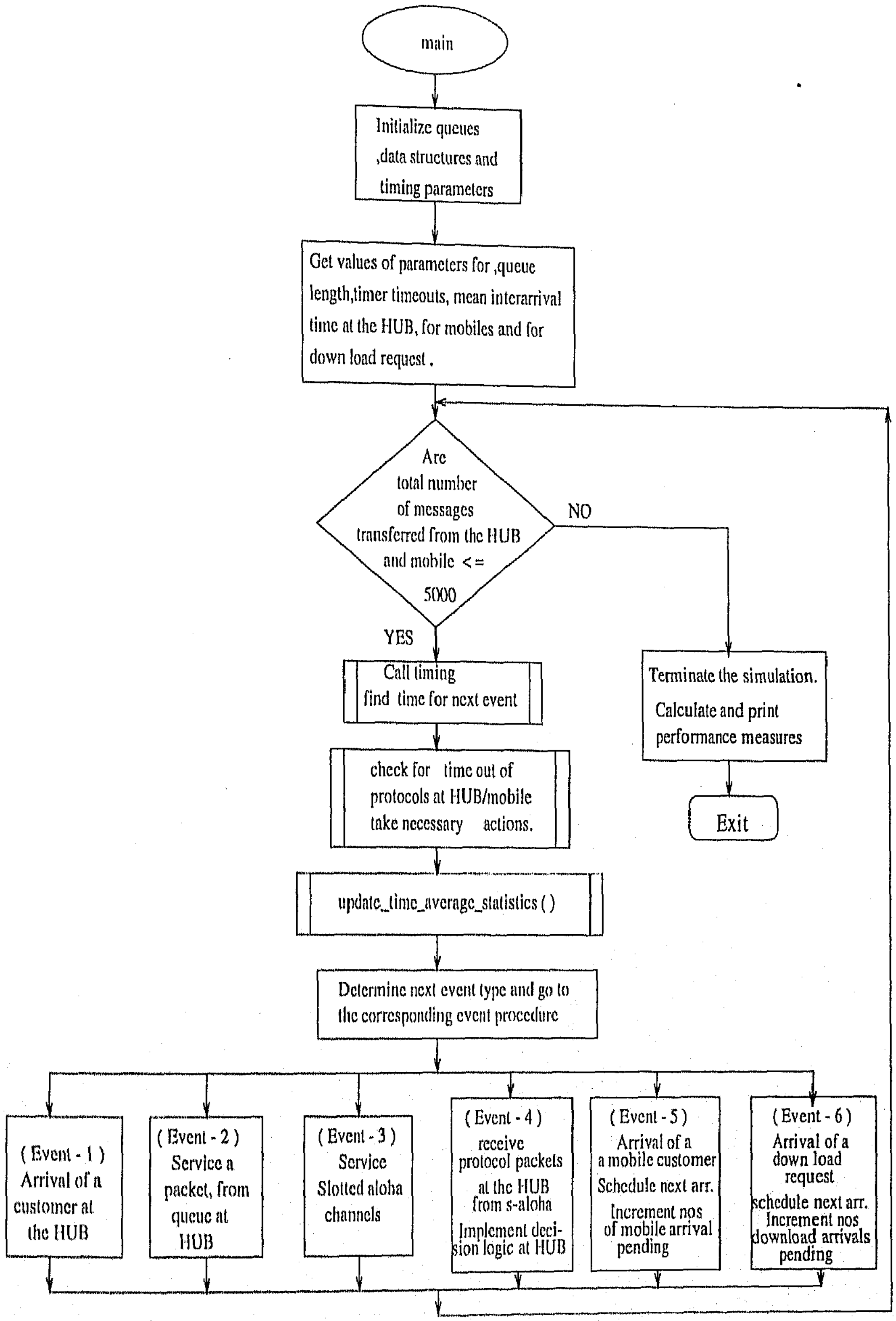
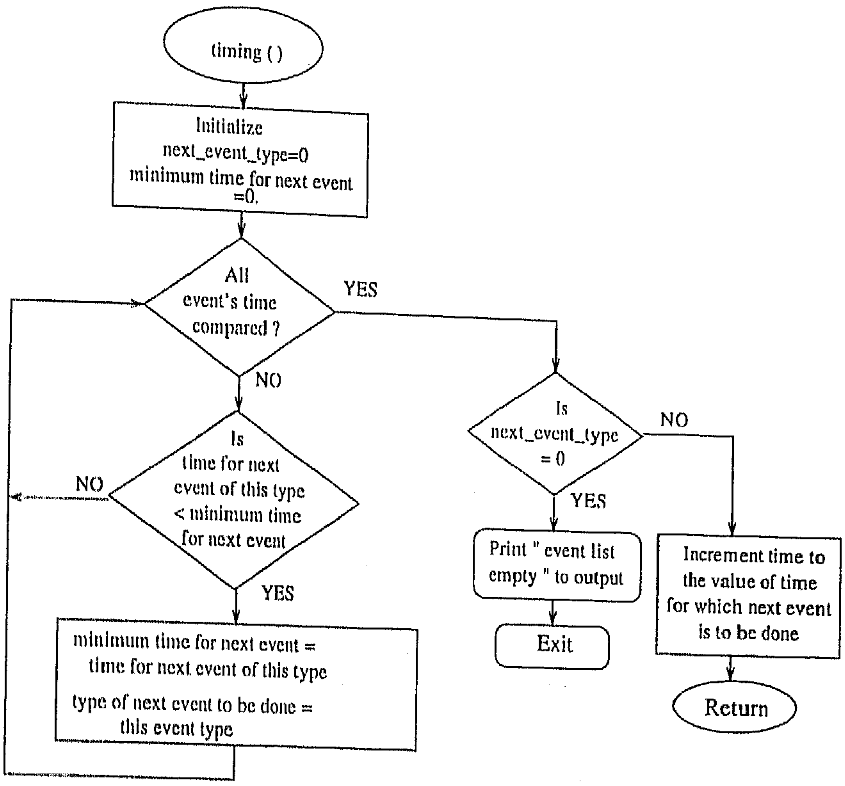
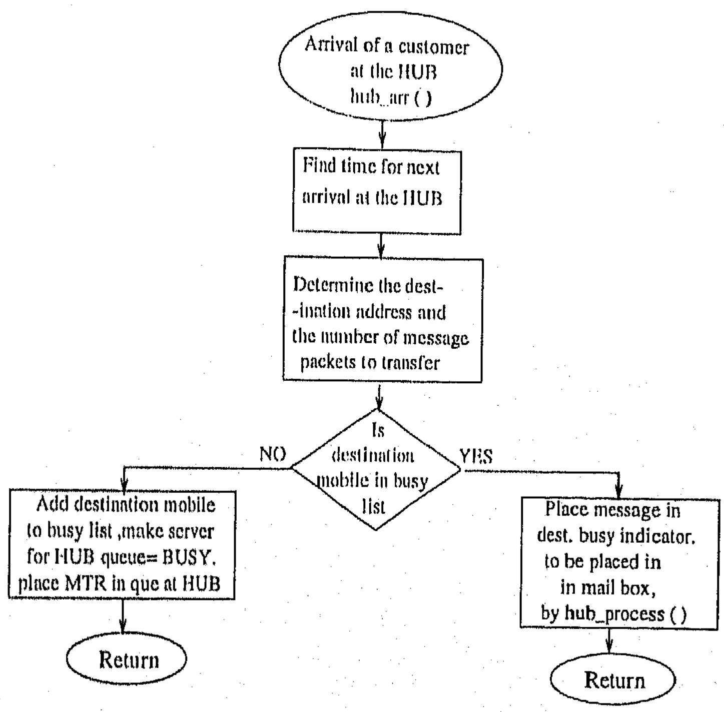
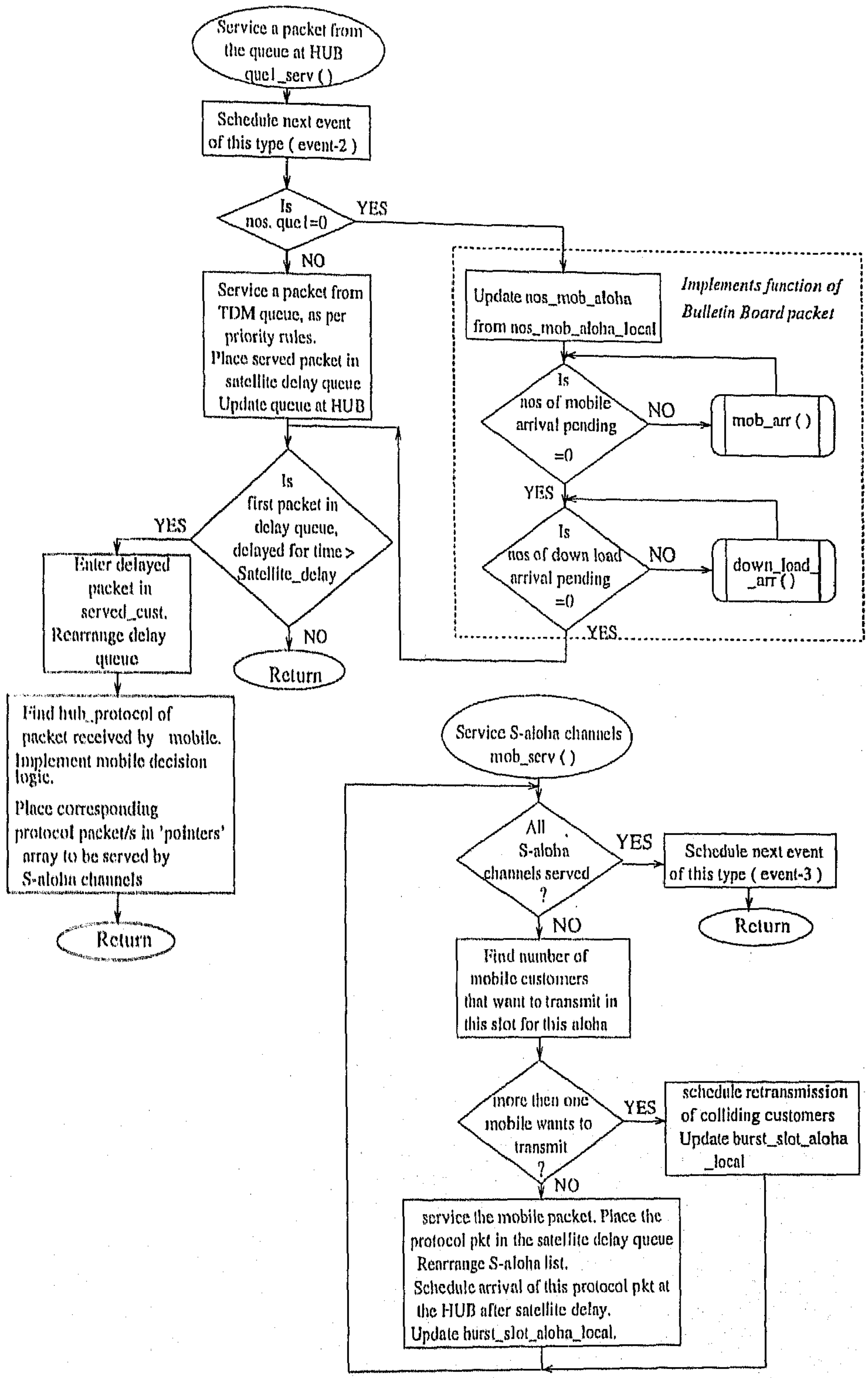


Figure 6.6: Flowcharts for simulator of Class-B type of system

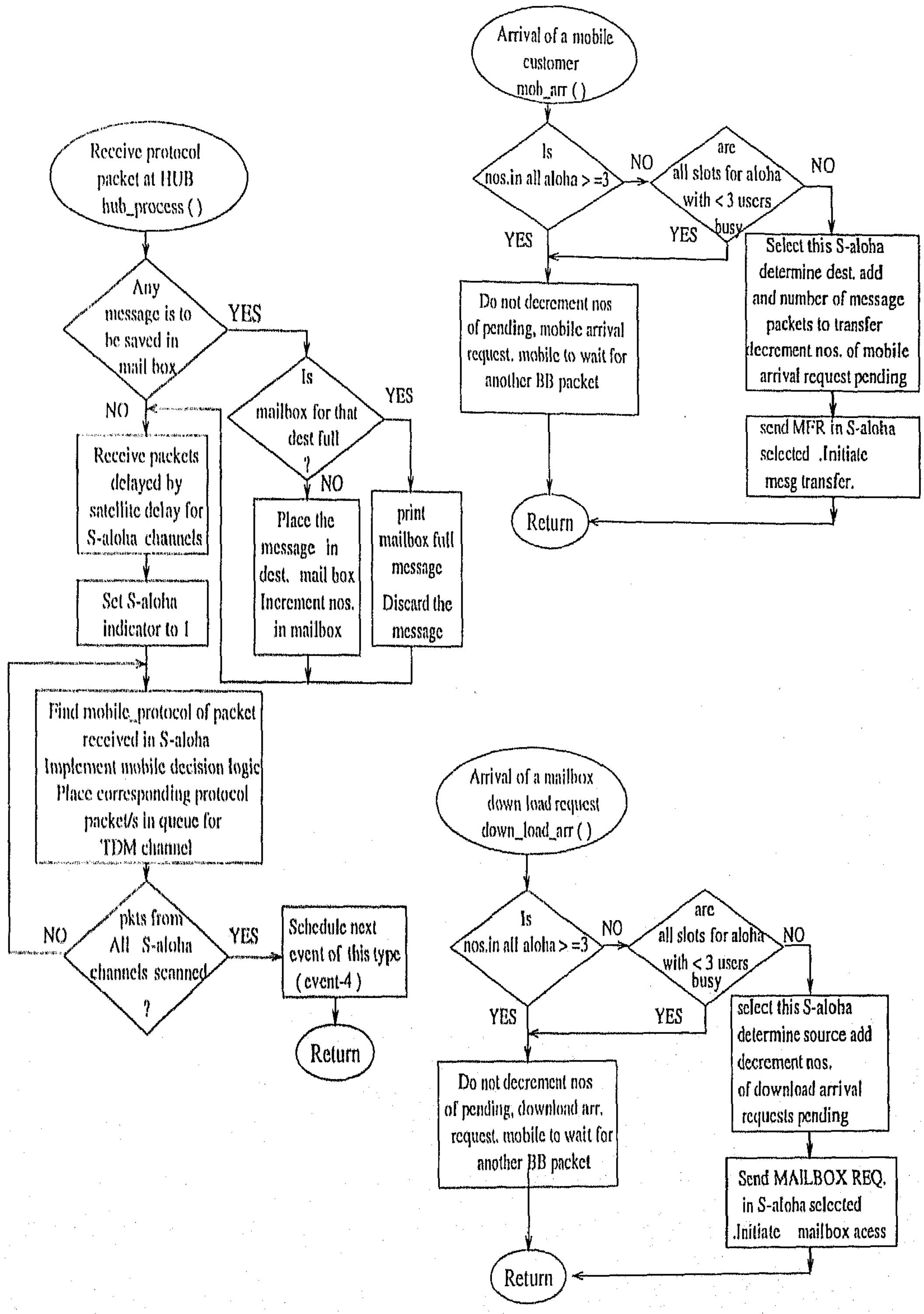




. . . Fig. 6.6 continued



. . . Fig. 6.6. continued



...Fig6.6. continued

6.4 Observations

As seen in the previous section, for operating the queues at the HUB and at mobile in steady state, the mean interarrival time at the HUB should be greater than 4.6 minutes and the mobile's mean interarrival time should be greater than 7.74 minutes. Based on these constraints, we restricted our study to interarrival times greater than 4.6 minutes at the HUB and 7.74 minutes for the mobile terminals. The confidence interval tests are conducted for some points in the performance plots. These points are represented by '*' in the performance plots and the corresponding values of the confidence intervals have been presented in Appendix A.

The average data transfer rate for HUB to mobile message transfer decreases steadily as the mean interarrival time at the HUB increases. This is seen in Fig. 6.7. This is typically what should be expected, since the HUB interarrival time considered in Fig. 6.7 is well above the limiting value of 4.6 minutes. The data transfer rate varies from 450 bits per second to a low of 300 bits per second. Different curves for three different values of mean interarrival time of the mobile customers have been shown in Fig. 6.7. These three curves do not show any significant correlation of HUB to mobile mean data transfer rate with the mean interarrival time of mobile customers. This is true only if the mean interarrival time for mobiles is greater than the limiting value. This observation can also be validated from Fig. 6.12 which shows that the average data transfer rate remains almost constant over the span of mobile mean interarrival time from 7.6 minutes to 8.5 minutes.

The variation of the mobile to HUB data transfer rate with respect to the mean interarrival time at the HUB is given in Fig. 6.8. There is a steady rise in the data transfer rate as the mean interarrival time at the HUB increases. This rises rapidly as the mean interarrival time at the HUB increases from 6 minutes to 6.5 minutes; beyond that, the data transfer rate increases relatively slowly.

This observation is contrary to expectations, since the acknowledgement packets for the mobile to HUB transfer are given higher priority on the TDM channel queue at the HUB than the data packets for the HUB to mobile transfer. This characteristic is observed because of the access requirement that a mobile must receive a Bulletin

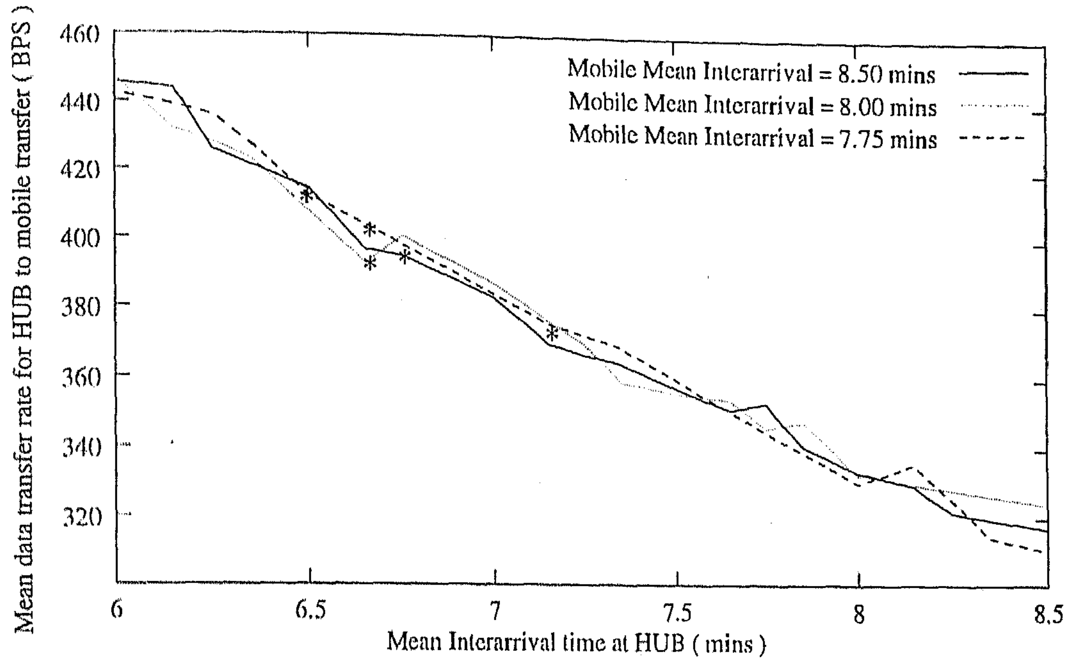


Figure 6.7: Variation of HUB to mobile data transfer rate with respect to HUB mean interarrival time (Down load mean interarrival time = 10 minutes)

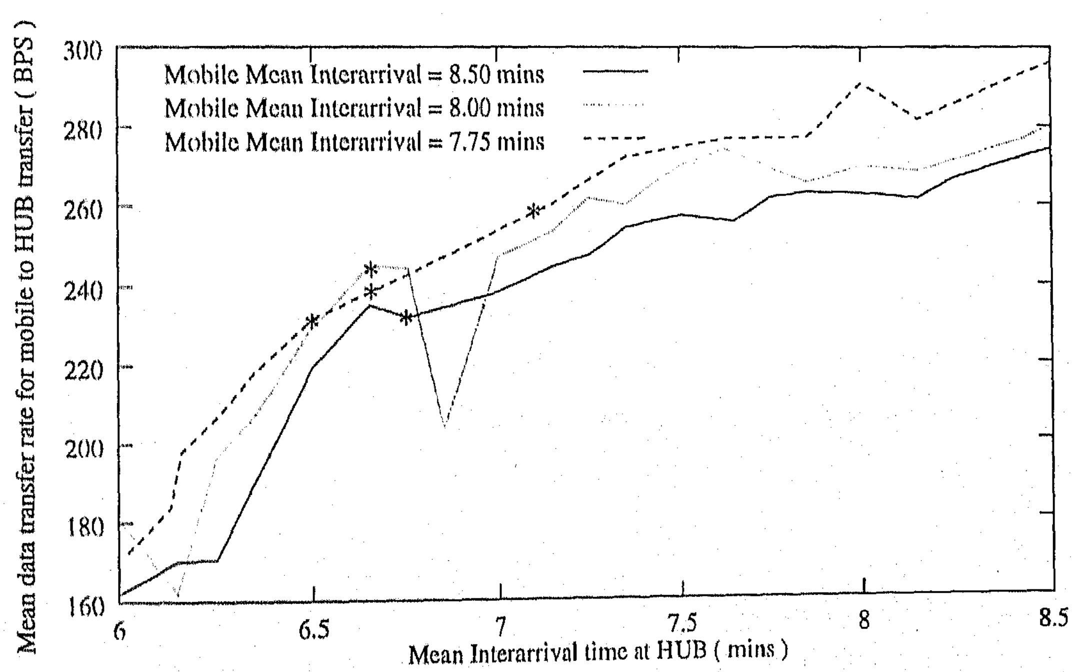


Figure 6.8: Variation of mobile to HUB data transfer rate with respect to HUB mean interarrival time (Down load mean interarrival time = 10 minutes)

Board packet, in order to choose a relatively free Slotted ALOHA channel. The Bulletin Board packet is transmitted with the least priority in the TDM channel. At lower values of HUB mean interarrival time, the time interval after which the BB packet gets transmitted is large. Hence the mobile terminals have to wait for a long duration before a BB packet is received and a message forward request can be transmitted. This results in less number of messages being transferred from the mobile to HUB which in turn gives a low mean data transfer rate. We call this as "BB delay effect". This phenomenon is more pronounced for the values of mean interarrival time less than 6.5 minutes and less predominant for the values greater than 6.5 minutes. This effect can also be verified by observing Fig. 6.9. As seen in Fig. 6.9, the average delay per data packet for mobile to HUB transfer decreases at a very low rate. The BB delay effect does not contribute to the delay per data packet. This because the average delay per data packet is calculated by measuring the time that elapses, since the Message Forward Request is sent and until a final acknowledgement of successful transfer of all the data packets is received. It can also be seen in Fig. 6.15, 6.16, 6.17, 6.18, that the throughput of Slotted ALOHA decreases significantly for lower values of mean interarrival time at HUB. This is due to the under-utilization of the Slotted ALOHA channels for the lower values of mean interarrival time, because of the BB delay effect.

From Fig. 6.8 we also find that the data transfer rate from mobile to the HUB increases with a decrease in mobile interarrival time. This observation is made for three different values of mobile mean interarrival time. This should be expected, since for lower values of mean interarrival time, the number of message transfers getting initiated are more. Moreover, for these values of interarrival time, the Slotted ALOHA channels are stable, resulting in higher data transfer rates. However, for lower values of mean interarrival time at the HUB, this advantage is lost due to the BB delay effect.

The average delay per data packet characteristics for HUB to mobile message transfer is shown in Fig. 6.10. The average delay decreases steadily for values of HUB mean interarrival time greater than 6.5 minutes; but there is a sharp rise in the value of average delay below this value. This observation signifies larger queuing delays for

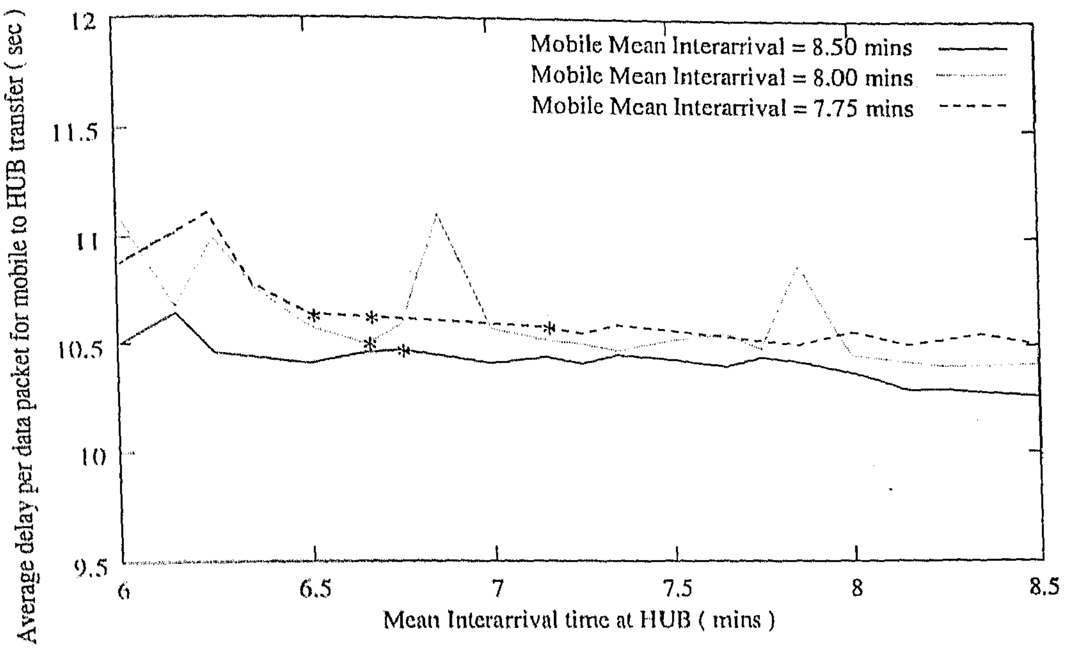


Figure 6.9: Variation of average delay per data packet for mobile to HUB message transfer with respect to HUB mean interarrival time (Down load mean interarrival time = 10mins)

values of mean interarrival time less than 6.5 minutes. Hence the BB delay effect is more pronounced for these values of mean interarrival time.

The average delay per data packet for the HUB to mobile data transfer is observed to be independent of mobile mean interarrival time. This observation can be made from Fig. 6.11, where it is seen that the average delay per data packet for HUB to mobile transfer remains almost constant for values of mobile interarrival times greater than the limiting value.

The mobile customers can also periodically send a request for checking their mailbox and to subsequently down load their mails. The mean interarrival time for down load request has been characterized as a parameter in Fig 6.13 and Fig. 6.14. It can be seen in Fig. 6.13 that the average data transfer rate, for HUB to mobile transfer is slightly better for the down load mean interarrival time of 10 minutes.

The comparative plots for performance of 4 parallel Slotted ALOHA channels for different traffic considerations are shown in Fig. 6.15, 6.16, 6.17, 6.18. Each of these

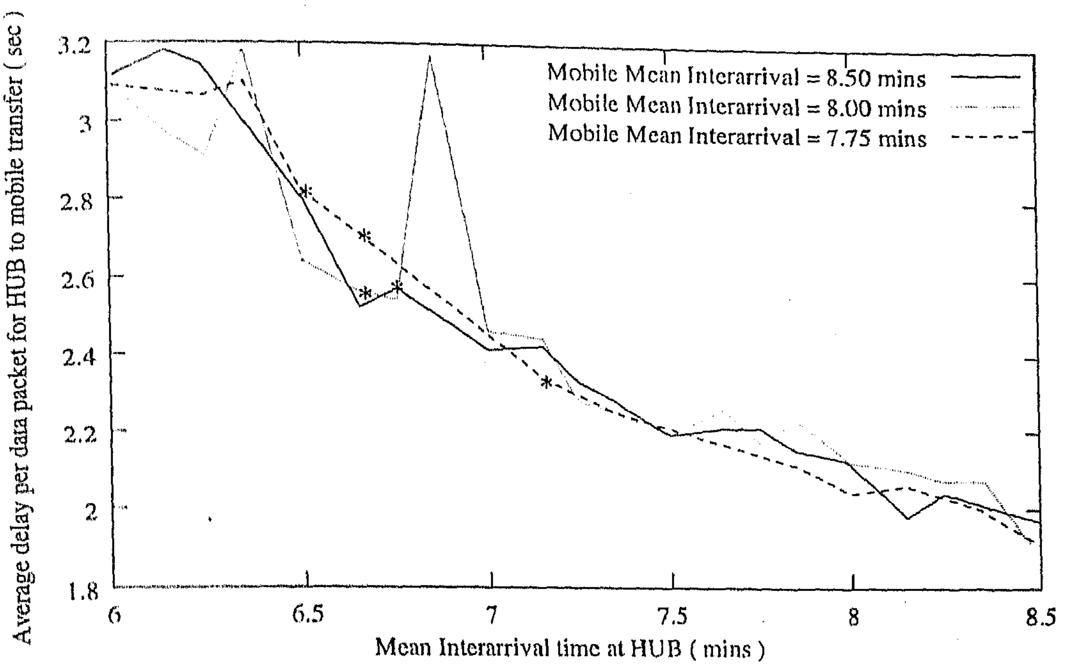


Figure 6.10: Variation of average delay per data packet for HUB to mobile message transfer with respect to HUB mean interarrival time (Down load mean interarrival time = 10mins)

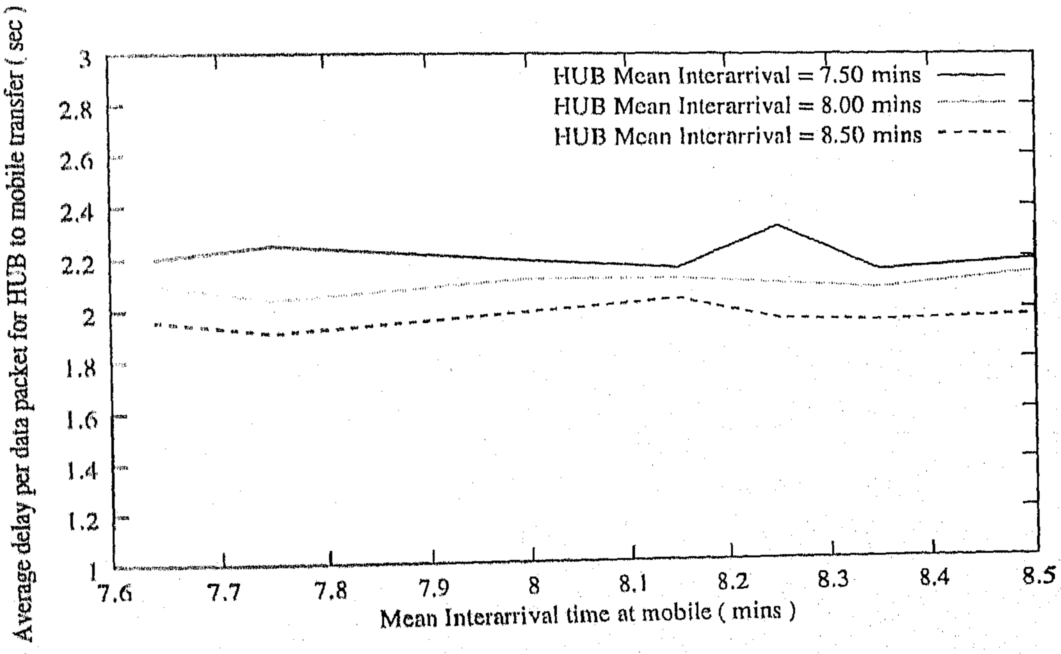


Figure 6.11: Variation of average delay per data packet for HUB to mobile message transfer with respect to mobile mean interarrival time (Down load mean interarrival time = 10mins)

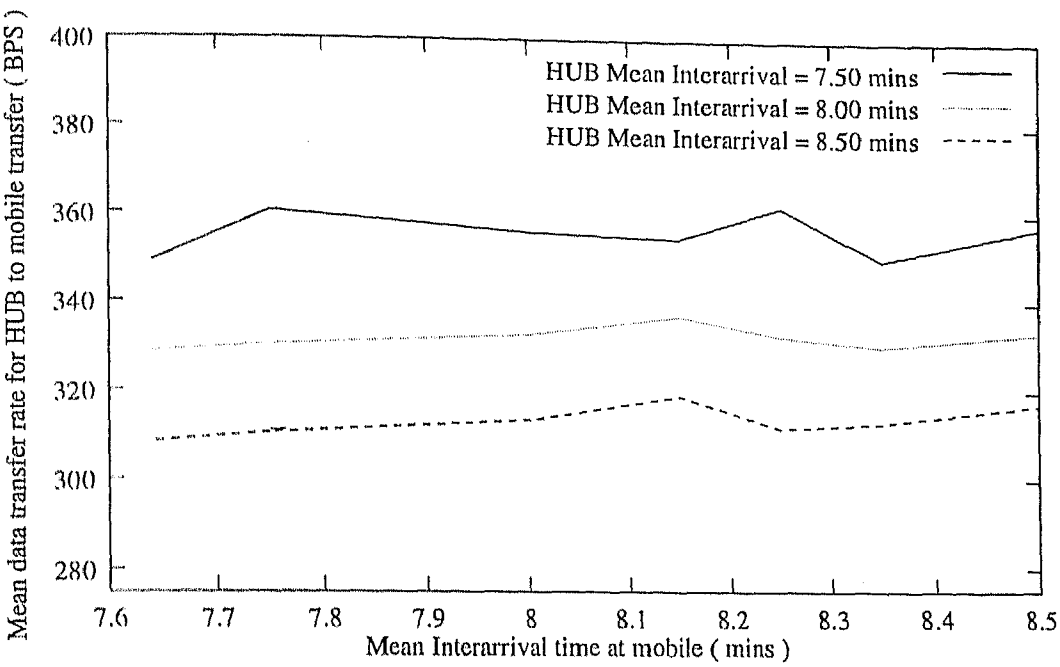


Figure 6.12: Variation of HUB to mobile mean data transfer rate with respect to mobile mean interarrival time (Down load mean interarrival time = 10mins)

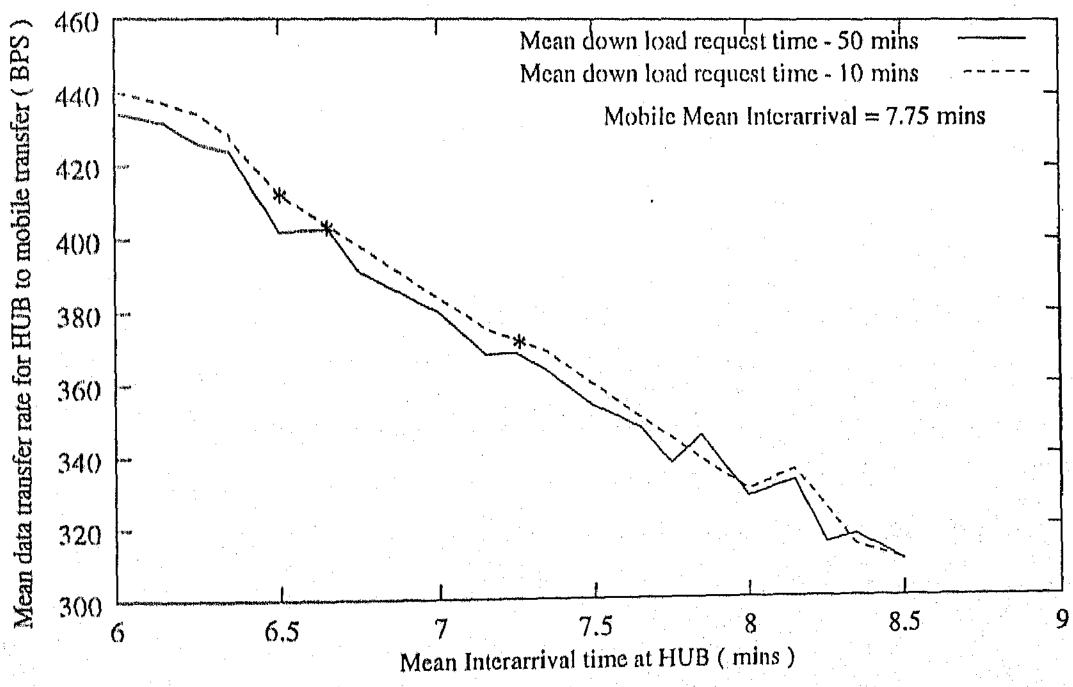


Figure 6.13: Effect on mean data transfer rate for HUB to mobile transfer due to change in mean down load interarrival time

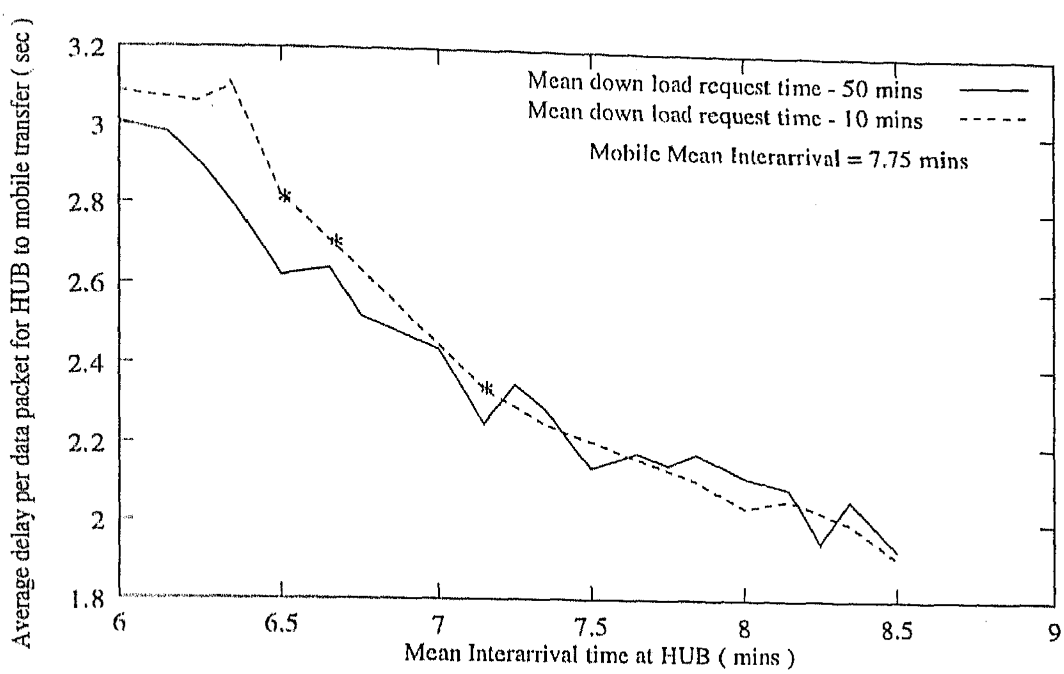


Figure 6.14: Effect on average delay per data packet for HUB to mobile transfer due to change in mean down load interarrival time

figures give variations of Slotted ALOHA throughput, as HUB mean interarrival time varies from 6 minutes to 8.5 minutes, for a fixed value of mobile mean interarrival time and mean down load interarrival time. Fig. 6.15 and 6.16 are plotted for mobile mean interarrival time of 8.50 minutes and for mean down load interarrival time of 10 minutes and 50 minutes respectively. It can be seen in the figure that the throughputs for all the four Slotted ALOHA channels increases sharply as the HUB mean interarrival time increase from 6.5 minutes to 7 minutes. This is due to the BB delay effect described earlier. Beyond 7 minutes, the throughput increases slowly with increase in HUB mean interarrival time; once again this slow rise is due to the decrease in traffic in TDM channel as mean interarrival time for the HUB increases. There is also a noticable difference in the characteristics of Fig. 6.15 and Fig. 6.16, for lower values of mean interarrival time at the HUB. The Slotted ALOHA channels are more stable and their throughput values are higher for the mean down load interarrival time of 50 minutes as compared that for 10 minutes. This is due to the reduced effect of BB delay for the mean down load interarrival time of 50 minutes. The data traffic from HUB to mobile, contributed with the mean down load interarrival time of 50 minutes is less than that contributed at the mean down load interarrival time

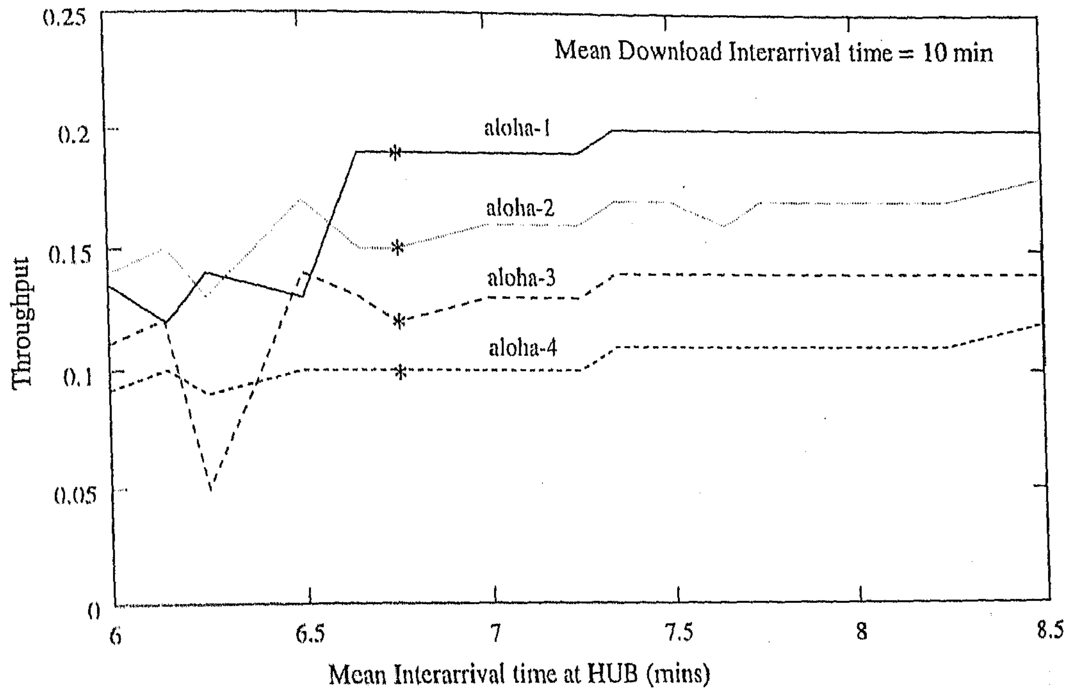


Figure 6.15: Performance of Slotted ALOHA (Mobile mean interarrival time = 8.5 minutes)

of 10 minutes. The same observation can also be made from Fig. 6.17 and Fig. 6.18 with the same reasoning.

The four Slotted ALOHA channels operate in parallel, but there is a difference in their throughput values; this variation is observed consistently in all the figures and for all values of HUB mean interarrival time. This nature of throughput characteristics for the Slotted ALOHA channels is because in the simulator, mobile customers select the Slotted ALOHA channels sequentially from aloha-1 to aloha-4; this results in Slotted ALOHA-1 getting the highest priority for any transfer of message to the HUB. This disparity can be balanced by selecting the Slotted ALOHA channels randomly.

The absolute values for throughput is dependent on the back-off mechanism to be used on detecting collisions in the Slotted ALOHA channel. It is therefore an implementation dependent aspect. In this simulation model, we have used a simple uniform back-off mechanism. The mobile terminal backs off after collision and tries to retransmit in one of the next 18 burst slots chosen randomly by a unformly distributed random number, which takes values between 1 to 18.

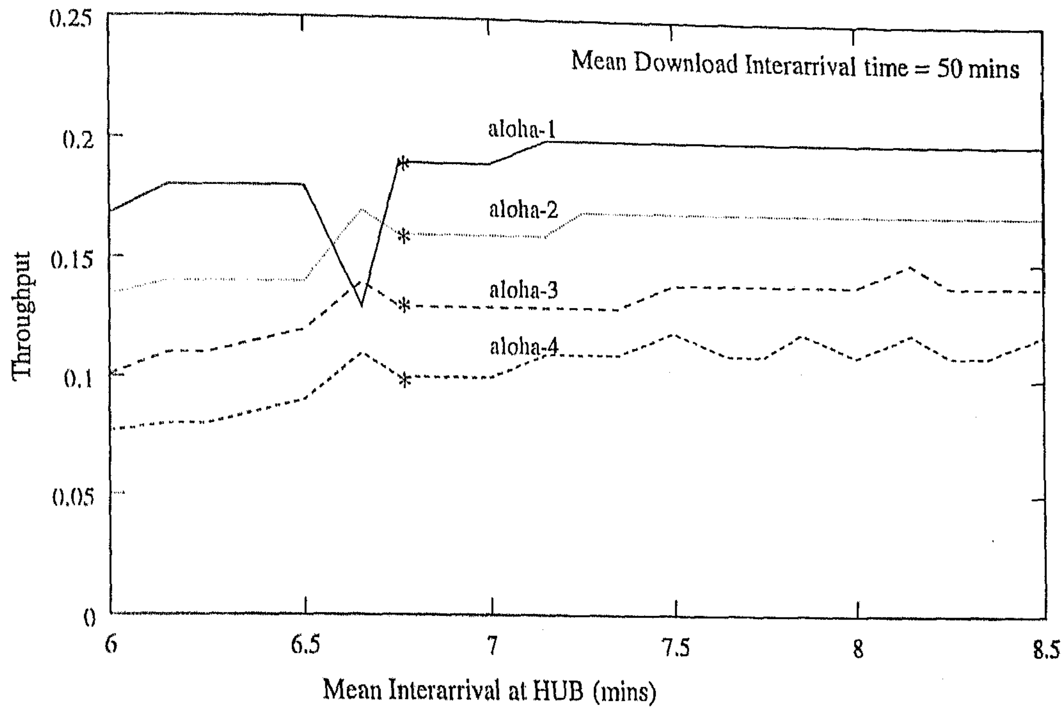


Figure 6.16: Performance of Slotted ALOHA (Mobile mean interarrival time = 8.5 minutes)

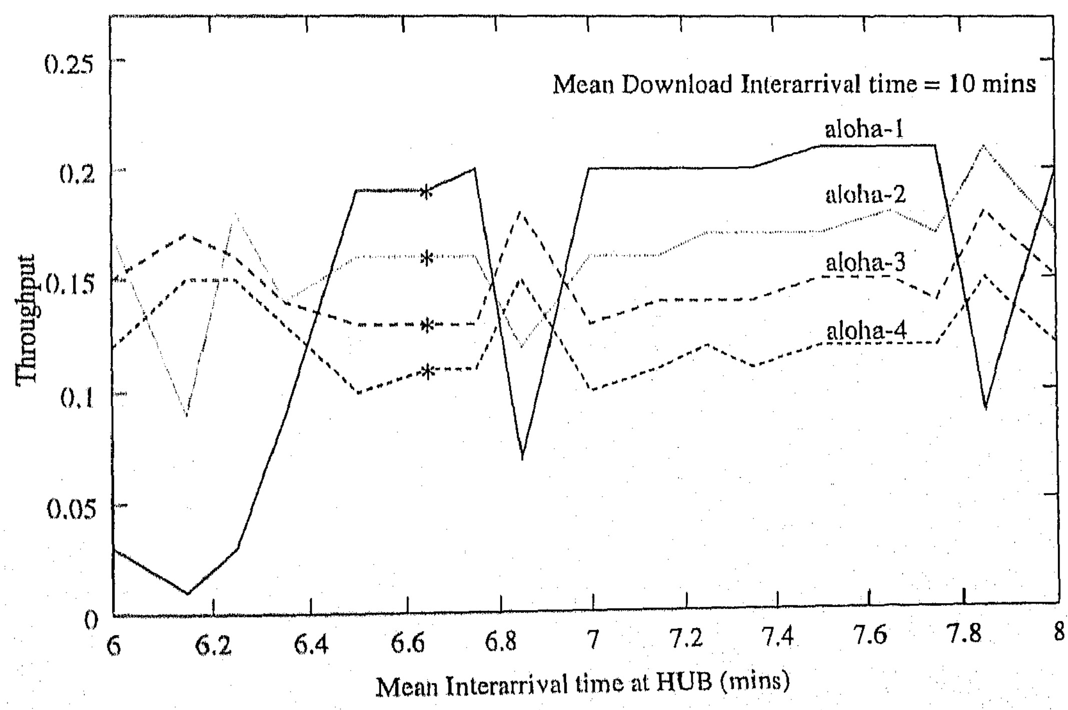


Figure 6.17: Performance of Slotted ALOHA (Mobile mean interarrival time = 8.00 minutes)

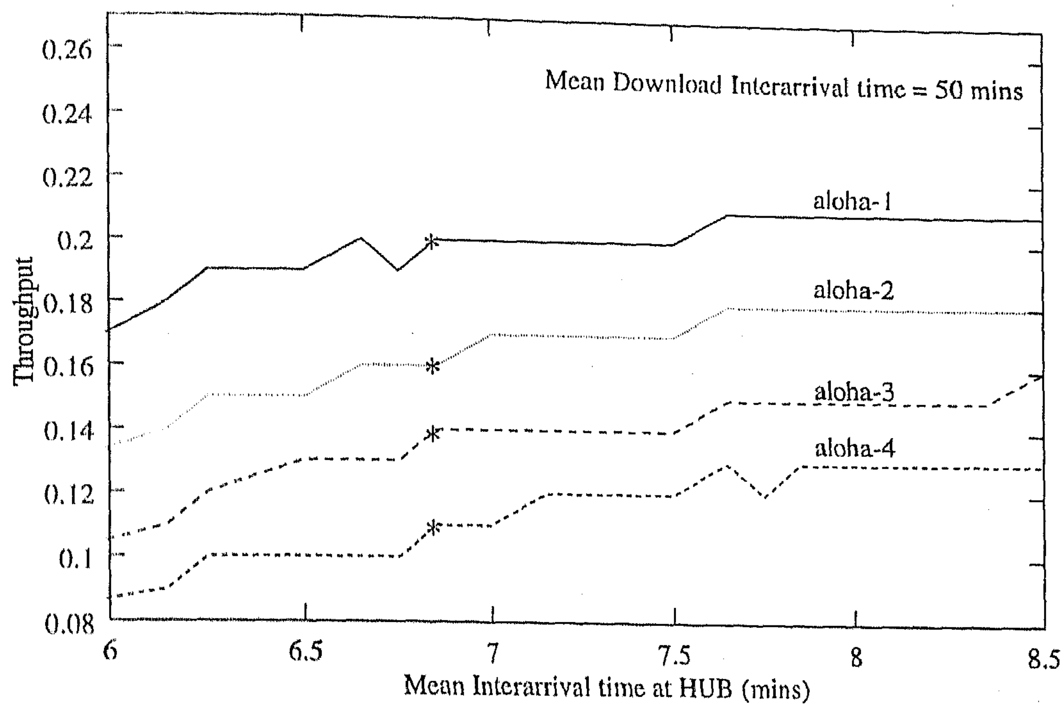


Figure 6.18: Performance of Slotted ALOHA (Mobile mean interarrival time = 8.00 minutes)

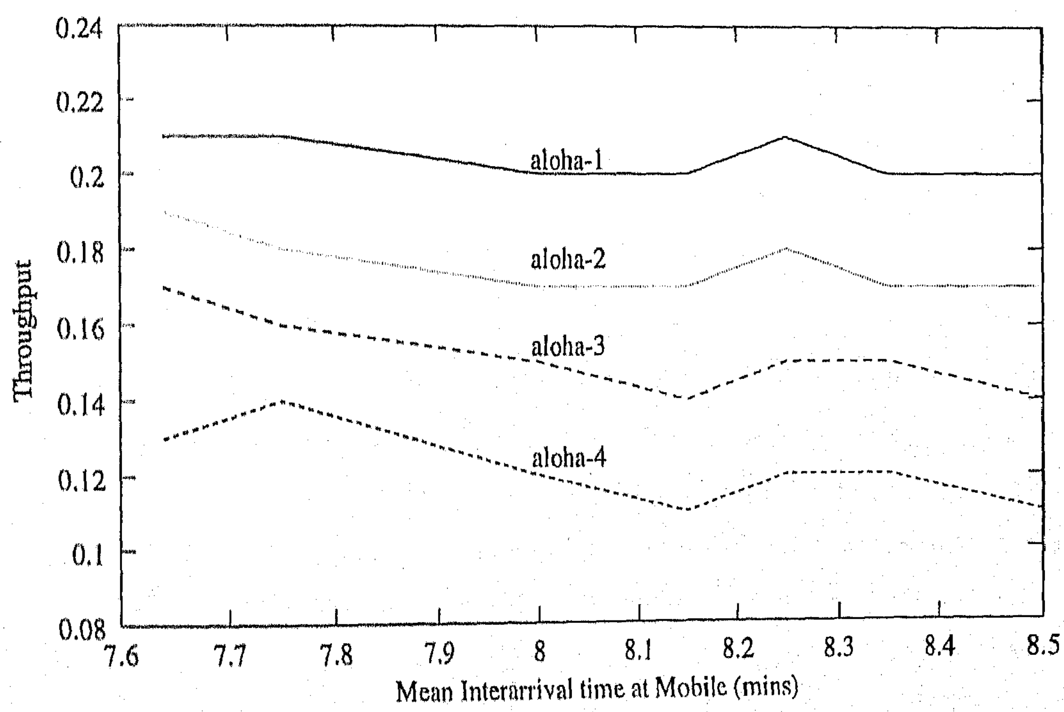


Figure 6.19: Performance of Slotted ALOHA (HUB mean interarrival time = 8.00 minutes)

6.5 Timers

The timers used in the system are simulated with their approximate values. These timers have been simulated as the time to live condition after the transmission of the protocol packet; the reply to this is awaited for resetting the corresponding timer. In this section, a brief discussion is presented on the characteristics of each timer along with their typical values and their overall effect on the steady state behavior of the system. The values of the timers have been chosen so as to obtain a reasonably low number of rejection of messages during the steady state operation of the system.

TIMEOUT_VAL_TWO : This is the value of the timer set at the HUB after the protocol packet MTR (02H) is placed in the TDM channel queue. The value used for the simulation for this timer is 17 minutes. The MTR packet has a lower priority in the TDM channel queue than the message packet (04H). Hence a high value for this timer should be used in order to get a reasonably low number of rejections of message transfer. A lower value of this timer leads to larger number of rejections of message transfer requests at low values of HUB mean interarrival time. This timer also performs the function of flow control at the message level. When the queuing delay for the MTR packet becomes large, the number of messages getting accepted for HUB to mobile transfer are reduced due to the larger number of rejections caused by the time out of this timer.

TIMEOUT_VAL_THREE: This is the value of the timer set at the mobile terminal after the protocol packet MTR ACK (03H) is transmitted by the mobile terminal. No formal parametric value has been specified for this timer. The mobile terminal is expected to wait for (N+10) TDM frame time for receiving the MRA request packet after the MTR ACK has been transmitted in the Slotted ALOHA channel; here N is the expected number of message packets. The timer value obtained in this way was found to give sufficiently low rejections for all conditions of traffic.

TIMEOUT_VAL_FOUR : This is the value of the timer set at the mobile terminal after transmitting all data packets. The mobile terminal waits for a MRA

packet before this timer is reset. A low value for this timer results in rejection of message transfer even if all the message packets have been transmitted. A relatively high value of this timer is used in the simulator. This is to avoid any loss of message transfer after the message packets has been transmitted. It is observed that the mobile terminal continues to transmit data packets, even though due to this timer the message transfer is aborted at the HUB. This is due to the lack of any feed back protocol packet indicating that the message transfer has been aborted . This results in wastage of the bandwidth of the Slotted ALOHA channels due to transfer of message packets which are subsequently rejected at the HUB.

TIMEOUT_VAL_FIVE : This is the value of the timer set at the mobile terminal after the MFR packet is transmitted by the mobile terminal. This timer is reset after the MFR ACK packet is received from the HUB. The value of this timer should not be greater than 30 seconds. Typically a very high value i.e. 5 minutes , has been used in the simulator to avoid any rejection of the message transfers from mobile to the HUB. This has been done since due to the BB delay effect, the number of message transfers initiated from the mobiles is observed to be less, this caused problems for the simulation to converge.

TIMEOUT_VAL_SIX : This is the value of the timer set at the HUB after the MFR ACK packet has been placed in the TDM channel queue at the HUB. This timer is reset after all the message packets are received at the HUB. The original protocol specifications mentioned that the HUB should wait for a time duration of (N+2) TDM frames after transmitting the MFR ACK, where N is the expected number of message packets. This condition has been observed to give a extremely high message rejections. Hence a fixed value i.e. TIMEOUT_VAL_SIX, has been added to the originally stated value of (N+2) TDM frames, as mentioned above. This constant value used in the simulator is 27 minutes.

TIMEOUT_VAL_SEVEN : This is the value of the timer set at the HUB after the MRA packet has been transmitted and the HUB waits for retransmission of the missing/errored packets, if any. The value used in the simulator is 8 minutes. No rejection of messages have been observed for this value of the timer.

TIMEOUT_VAL_EIGHT : This is the value of the timer set at the HUB after the MRA request (08H) is placed in it's queue for the TDM channel. This prevents infinite waits at the HUB for the MRA packet. In the simulator, this timer was set after the MRA request packet has been placed in the HUB's queue. This is done at the same time as placing the message packets in the TDM channel queue. Since all the message packets will then have to be transmitted before the transmission of the MRA request packet, this time value has been chosen to be rather high in the simulator i.e. 17 minutes.

6.6 Results

In the previous sections we studied the behavior of the Class-B type of system under various conditions of traffic. Various performance characteristics of the system have been plotted and the reasons for their behavior are stated. The access control scheme for the Slotted ALOHA channels, followed in the system gives adequate mechanism for flow control for mobile originated transfers. The timers associated with the message transfer requests, for data transfer from the mobile to HUB also contribute to the flow control at message level.

The main constraint in the system has been observed due to the BB delay effect as described in Section 6.6. The drawbacks faced because of the transmission of BB packets only when the TDM queue at the HUB is empty can be overcome by a small change in this scheme. The BB packets can be transmitted at regular intervals, instead of being transmitted only when there are no other protocol packets to be transmitted in the TDM channel queue. This ensures a regular feedback of Slotted ALOHA activity data to the mobile terminals; consequently, the mobiles do not have to wait for a long duration for selecting a Slotted ALOHA channel. This scheme gives much better performance of average data transfer rate for mobile to HUB transfer. There is a significant improvement in average data transfer rate for lower values of HUB mean interarrival time, for which the effect of BB delay has been observed to be more pronounced. These inferences may be drawn from Fig. 6.20.

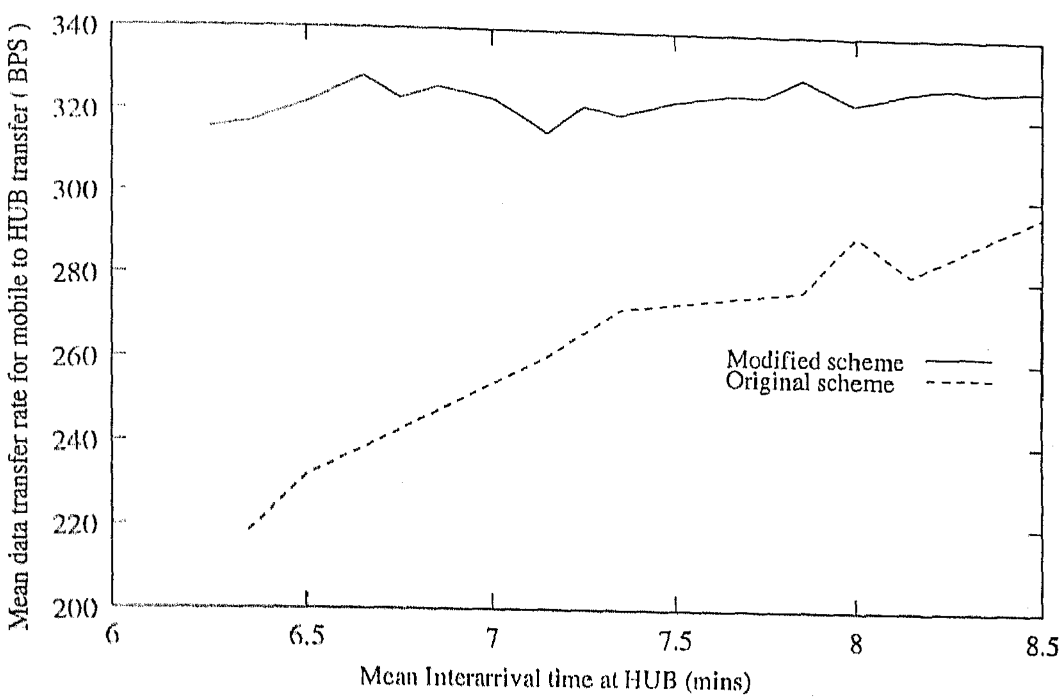


Figure 6.20: Effect of change in BB transmission scheme on mean data transfer rate for mobile to HUB transfer

shows the original scheme and the modified scheme with 4.64 minutes as the time period between transmission of two successive BB packets. The BB packet will also be transmitted when there are no other protocol packets present in the TDM channel queue. With this modified scheme the overhead for transmission of BB packets is less than 0.4% of the bandwidth for TDM channel at HUB.

The effect of this scheme for other parameters is also shown in Fig. 6.21, 6.22, 6.23 and 6.24. Fig. 6.21 shows an increase in the average data transfer rate for HUB to mobile transfer for the modified scheme. This is due to the larger number of transfer of mails to mobile terminals; this also results because of reduced BB delay effect, better access is available to mobiles requesting for down loading from the mailbox. The difference in the HUB to mobile data transfer rate reduces for higher values of HUB mean interarrival time. This can be observed in Fig. 6.21 for the two curves. This is because, less number of mails are collected in the mailboxes at higher values of mean interarrival time as there are fewer rejections caused by the timeouts. It can also be seen in Fig. 6.22., that for the modified scheme, the average delay per data packet for HUB to mobile transfer also increases due to larger queuing delays

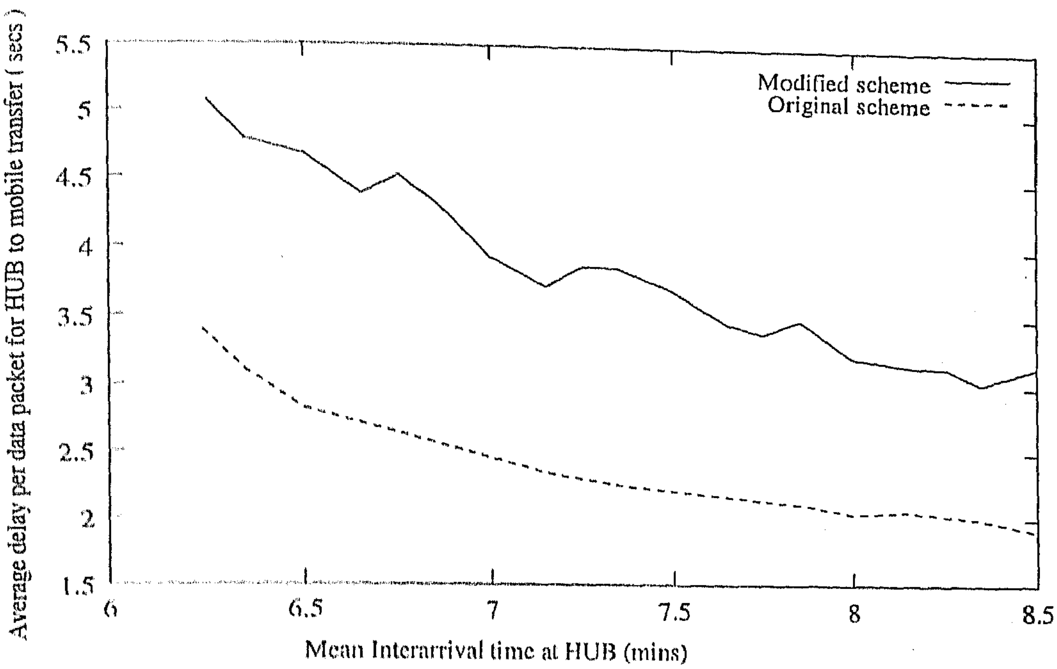


Figure 6.22: Effect of change in BB transmission scheme on average delay per data packet for HUB to mobile transfer

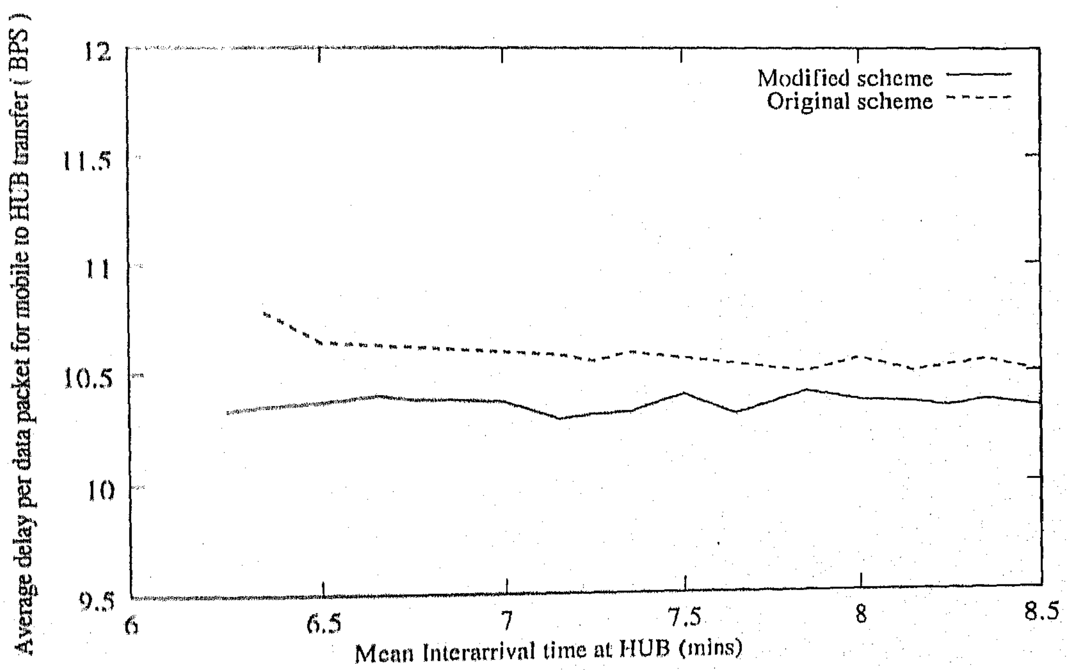


Figure 6.23: Effect of change in BB transmission scheme on average delay per data packet for mobile to HUB transfer

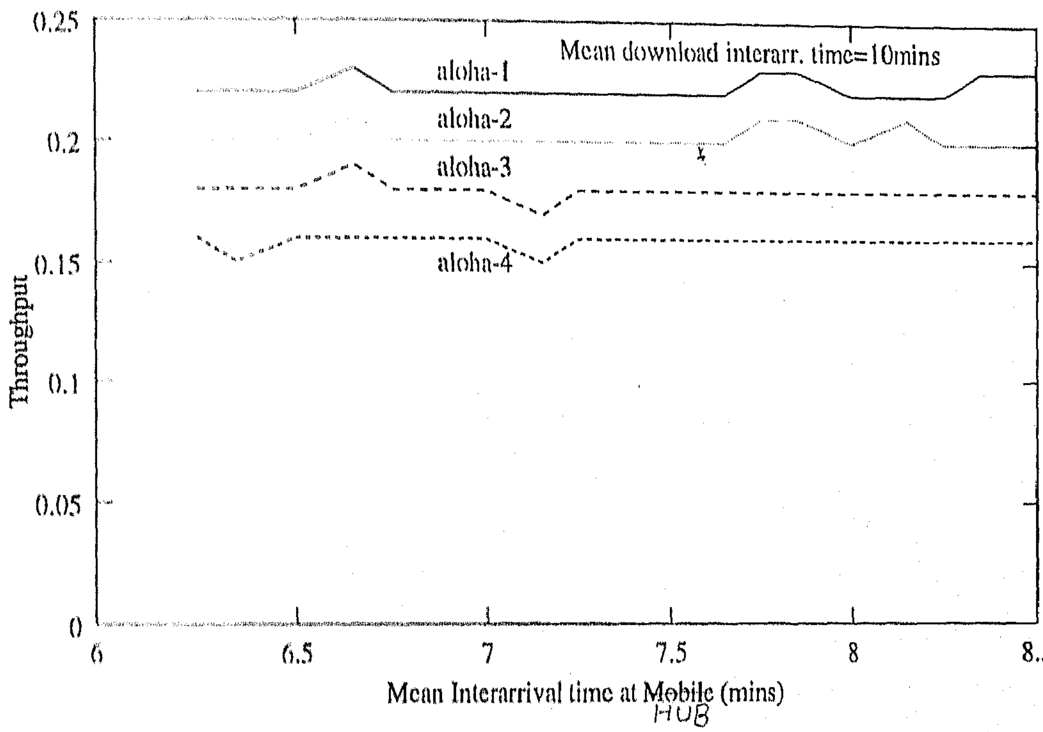


Figure 6.24: Slotted ALOHA performance for mobile mean interarrival time = 7.75 minutes with modified BB transmission scheme

Chapter 7

CONCLUSION AND SUGGESTED CHANGES

7.1 Conclusions

The performance characteristics presented in Chapter 4. and Chapter 6., give some idea of the performance of the system. The Class-A type of system is found to be capable of supporting reasonably high traffic rate with low blocking probability for number of SCPC channel pairs greater than 100. The speech codec to be used for voice transmissions does make voice communication possible and economical using 8 kbps SCPC channels. This system may be used for higher traffic intensity, provided the bottle neck due to unsatisfactory S-ALOHA throughput may be overcome.

The Class-B system performance is found to be adequate for a low bit rate store and forward message transfer. The protocols to be used for this system were verified through simulation. They are found to perform adequately. The access control mechanism is found to provide adequate flow control at the message level; but at the same time this scheme is found to starve the mobiles for access. In the next section we suggest some modifications to be made in order to further enhance the performance of this system.

7.2 Suggested changes

1. The BB transmission scheme was found to perform unsatisfactorily in the Class-B type of system. This scheme can be effectively modified as suggested in the previous chapter.

2. For the HUB to mobile message transfer, the HUB is expected to wait for (N+2) TDM frame times after the transmission the of MFR ACK packet for receiving all data packets from the mobile. This value should be increased to ensure successful transfer of messages from mobile to HUB even if there are relatively larger delays in the Slotted ALOHA channel for the data packets. A discussion for the possible value of timeout to be used for this timer i.e. TIMEOUT_VAL_SIX has been given in the previous chapter.

3. For the above mentioned timer, the mobile terminal continues to transmit the data packets (04H), even though due to time out at the HUB, these packets would be subsequently rejected. In this manner the mobile terminals add to the wastage of bandwidth by transmitting packets which prove to be redundant at the HUB. At the same time it also blocks access to these Slotted ALOHA channels for other mobile terminals which may be waiting to obtain a relatively free Slotted ALOHA channel. This drawback can be overcome by introducing a new protocol packet from the HUB, to be transmitted with a high priority, that will communicate the rejection of message transfer to the concerned mobile terminal. This "Transfer status" packet can be allocated any one of the spare packet types.

4. The "Transfer status" packet can be similarly used to communicate the rejection of message transfer at the mobile terminal due to the timeout of the timer that is set up after the transmission of the MTR ACK packet. The reception of this packet at the HUB will stop the transmission of data packets which are subsequently lost at the mobiles.

5. For improving the performance of Slotted ALOHA channel a divergence from a simple Slotted ALOHA scheme towards a more deterministic multiaccess scheme can be made. The requirement that only three mobile terminals can transmit

at the same time in any Slotted ALOHA channel can be further extended to encompass each slot in any given frame. There are 6 burst slots in any given TDM frame duration. A transmitting mobile terminal can be allocated two of the given six burst slots in a frame to transmit their protocol packets. One of the slot from these two slots can then be randomly chosen by the mobile terminal for transmission. The HUB shall maintain the data for allocation of pairs of burst slots for each Slotted ALOHA. The allocation can be done in the MFR ACK packet. The bytes left as spare in the MFR ACK packet [4] can carry the bit pattern to identify the pair of bursts that the mobile should select. This scheme will also preserve the multiaccess feature for the return channel, providing adequate mechanism for the mobiles to transmit their protocol packets for the HUB to mobile transfer.

Appendix A

TABLES FOR CONFIDENCE-INTERVAL TEST

Results of Confidence Interval tests are tabulated in the following eight tables. The tables include the values of the sample mean, the sample variance, the upper confidence-interval endpoint and the lower confidence-interval end point. The confidence intervals are created for 95% confidence for respective performance parameter values.

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	HUB to Mobile Mean Data Transfer Rate			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7 15	7 75	10	374 424	42 170	377 464	371 385
2	6 5	7 75	10	412 278	34 580	415 113	409 444
3	6 65	7 75	10	401 690	24 305	403 997	399 382
4	6 65	8 00	10	401 475	22 148	403 677	399 272
5	6 75	8 50	10	398 131	31 386	400 831	395 431
6	6 75	8 50	50	397 310	27 342	399 830	394 790

Table for 95% confidence interval for HUB to mobile mean data transfer rate

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	Mobile to HUB Mean Data Transfer Rate			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7 15	7 75	10	259 850	88 516	264 253	255 446
2	6 5	7 75	10	229 300	256 629	237 021	221 578
3	6 65	7 75	10	239 404	232 656	246 543	232 266
4	6 65	8 00	10	235 710	157 256	241 578	229 841
5	6 75	8 50	10	227 568	145 000	233 372	221 764
6	6 75	8 50	50	234 950	25 4699	237 311	232 588

Table for 95% confidence interarrival for mobile to HUB mean data transfer rate

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	Average Delay Per Data pkt (HUB to Mobile trans)			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7 15	7 75	10	2 388	0 0183	2 451	2 324
2	6 5	7 75	10	2 775	0 0297	2 858	2 692
3	6 65	7 75	10	2 650	0 0173	2 712	2 588
4	6 65	8 00	10	2 617	0 0044	2 648	2 585
5	6 75	8 50	10	2 560	0 0112	2 611	2 508
6	6 75	8 50	50	2 581	0 0045	2 612	2 550

Table for 95% confidence interarrival for average delay per data packet for HUB to mobile transfer

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	Average Delay Per Data Pkt (Mobile to HUB Trans)			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7 15	7 75	10	10 652	0 0335	10 737	10 566
2	6 5	7 75	10	10 751	0 0689	10 878	10 625
3	6 65	7 75	10	10 717	0 0442	10 815	10 619
4	6 65	8 00	10	10 588	0 0072	10 628	10 548
5	6 75	8 50	10	10 567	0 0348	10 657	10 477
6	6 75	8 50	50	10 429	0 0016	10 448	10 409

Table for 95% confidence interarrival for average delay per data packet for the mobile to HUB transfer

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	Slotted ALOHA 1 Throughput			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7.15	7.75	10	0.175	0.002993	0.201	0.149
2	6.5	7.75	10	0.141	0.004953	0.175	0.107
3	6.65	7.75	10	0.154	0.001008	0.184	0.124
4	6.65	8.00	10	0.166	0.001762	0.185	0.146
5	6.75	8.50	10	0.132	0.003717	0.161	0.103
6	6.75	8.50	10	0.189	0.000006	0.190	0.188

Table for 95% confidence interarrival for Slotted ALOHA 1 throughput

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	Slotted ALOHA 2 Throughput			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7.15	7.75	10	0.179	0.000263	0.187	0.172
2	6.5	7.75	10	0.161	0.000482	0.171	0.150
3	6.65	7.75	10	0.169	0.000194	0.175	0.162
4	6.65	8.00	10	0.155	0.001291	0.172	0.138
5	6.75	8.50	10	0.164	0.000811	0.178	0.151
6	6.75	8.50	50	0.153	0.000013	0.155	0.151

Table for confidence interarrival for Slotted ALOHA 2 throughput

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	Slotted ALOHA 3 (Throughput)			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7.15	7.75	10	0.152	0.000262	0.159	0.144
2	6.5	7.75	10	0.145	0.000539	0.156	0.133
3	6.65	7.75	10	0.144	0.000288	0.152	0.136
4	6.65	8.00	10	0.136	0.000224	0.143	0.129
5	6.75	8.50	10	0.138	0.000491	0.149	0.128
6	6.75	8.50	50	0.124	0.000009	0.125	0.122

Table for 95% confidence interarrival for Slotted ALOHA 3 throughput

	HUB Mean Interarr Time (mins)	Mobile Mean Interarr Time (mins)	Mean Down Load Interarr Time (mins)	Slotted ALOHA 4 (Throughput)			
				Sample Mean	Sample Variance	Upper Confidence Interval Point	Lower Confidence Interval Point
1	7.15	7.75	10	0.125	0.000238	0.132	0.172
2	6.5	7.75	10	0.118	0.000495	0.129	0.108
3	6.65	7.75	10	0.121	0.000445	0.130	0.111
4	6.65	8.00	10	0.116	0.000527	0.127	0.106
5	6.75	8.50	10	0.116	0.000538	0.127	0.105
6	6.75	8.50	50	0.097	0.000009	0.099	0.096

Table for 95% confidence interval for Slotted ALOHA 4 throughput

Appendix B

APPROXIMATE ANALYSIS FOR CLASS-B SYSTEM

λ_1 Arrival rate of customers at the HUB

λ_2 Arrival rate of mobile customers

$\lambda_2' = 3\lambda_2$ Arrival rate of MIR ACK packets

$\lambda_1' = \lambda_1$ Arrival rate of MIR ACK packets in S ALOHA channels

λ_2'' $3\lambda_2$ Arrival rate of MRA packets at the HUB

λ_1'' λ_1 Arrival rate of MRA packets in S ALOHA channels

m_1 Average number of packets retransmitted from the HUB,
due to loss of packets for a HUB to mobile transfer

m_1 = (Average number of data packets to be retransmitted
per message transfer from HUB to mobile) $+1$

m_1' = Average number of packets retransmitted in the S ALOHA channels,
due to loss of packets for a HUB to mobile transfer

$m_1' = 1$ {due to retransmission of MRA packet}

m_1 = Average number of packets retransmitted in the S-ALOHA channels, due to loss of packets for a mobile to HUB transfer

m_2 = (Average number of data packets to be retransmitted per message transfer from mobile to HUB)

m_2' = Average number of packets retransmitted from the HUB, due to loss of packets for a mobile to HUB transfer

m_2' = 3 {due to retransmission of MRA packets}

prob(first retransmission of message packets) = $p_1 = \{1-(1-\text{packet error prob})^{250}\}$

also p = packet error prob = $\{1-(1-\text{bit error prob})^{640}\}$

where bit error prob = 1×10^{-6}

Hence $p = 6.39 \times 10^{-4}$

Hence $p_1 = 0.14785$

Therefore the average number of data packets to be retransmitted

$$\frac{1}{100}\sum_{N=1}^{100}\sum_{k=1}^{N} k \binom{N}{k}(p)^k(1-p)^{N-k}$$

$N_{retrans} = 0.16$

The total arrival rate at the TDM queue can therefore be written as

$$\lambda_{HUB} = 250\lambda_1 + 2\lambda_1 + \lambda_2' + \lambda_2'' + p_1(m_1\lambda_1 + m_2'\lambda_2)$$

$$= 6\lambda_2 + 252\lambda_1 + p_1(m_1\lambda_1 + m_2'\lambda_2)$$

therefore substituting the values of m_1, m_2' and p_1

$$\lambda_{HUB} = \{0.147(N_{retrans} + 1) + 252\}\lambda_1 + \{3 \times 0.147 + 6\}\lambda_2$$

$$= 6.443\lambda_2 + \{0.147(N_{retrans} + 1) + 252\}\lambda_1$$

let μ_{HUB} = mean service rate of protocol packets at the HUB

Now one TDM frame (i.e. 8 slots) is of 8.64 sec. Hence average service time for each protocol packet = 1.08 sec

therefore we can write $\mu_{tdm} = \frac{1}{1.08}$ per second

Now the condition for stable operation of the TDM queue can be written as

$$\frac{\lambda_{tdm}}{\mu_{tdm}} < 1$$

therefore it can be written as

$$\frac{1}{1.08} > 6.11\lambda_2 + 252.17\lambda_1 \qquad \text{(B.1)}$$

Similarly the analysis for the S-ALOHA system can be done for it's stable operation.

The total arrival rate to the S-ALOHA system can therefore be written as

$\lambda_{aloha} = 250\lambda_2 + \lambda_2 + \lambda_1' + \lambda_1'' + p_1(m_1'\lambda_1 + m_2\lambda_2)$

Substituting the values and solving this equation we get

$\lambda_{aloha} = 2.11\lambda_1 + (251 + 0.14N_{retrans})\lambda_2$

therefore $\lambda_{aloha} = 2.11\lambda_1 + 251.02\lambda_2$

let μ_{aloha} = mean service rate of protocol packets from the S-ALOHA system

During one TDM frame of 8.64 sec, six burst slots are present per S-ALOHA.
Hence average service rate for each S-ALOHA

$= \frac{1}{1.44} \times$ (S-ALOHA throughput)

Assuming that throughput for each S-ALOHA is 0.2 and summing the service rates for all four S-ALOHA channels, we obtain the mean service rate from the S-ALOHA system

i.e. $\mu_{aloha} = 0.55$ per sec

The condition for stable operation of the S-ALOHA system can be written as

Therefore we can write the above equation as

$$0.55 = 2.11\lambda_1 + 25.102\lambda_2 \tag{B 2}$$

Considering only the equalities for the Eqn. B.1 and Eqn. B.2 and solving them we obtain

$$\lambda = \frac{1}{[illegible]} \text{ per minute}$$

$$\lambda_1 = \frac{1}{[illegible]} \text{ per minute}$$

i.e. mean interarrival time at the HUB = 1.6 minutes

and mean interarrival time of the mobiles = 7.74 minutes

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